Hi-Fi Troubles

Herman Burstein

...how *you* can avoid them

...how *you* can cure them

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How you can cure them

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With a specially written chapter for the guidance of the English reader by W. Oliver

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It is essential that the English reader should read this chapter.

"Hi-fi" and "trouble" are rather incompatible terms! Indeed, you might say that when troubles come into audio equipment, high fidelity departs from it and doesn't return till the troubles have been cleared.

This book will help you to avoid audio troubles arising in the first instance; and will help you to get rid of them as quickly as possible if they do occur.

As it is a book of American origin, a few points in it may need re-orientating when using it in Britain. We will start by looking at these briefly, in the order in which you will come across them when reading the text and looking at the illustrations.

Page 7: "tubes" means valves. P. 15: vacuum-tube voltmeter is the American term for valve voltmeter; VTVM is the abbreviation. P. 16: 60-cycle current (60 Hertz or Hz.) is the type of supply on American domestic mains; the voltage is around 110–120, often expressed as some nominal figure such as 115 or 117. Like ours, the current is AC, but our voltage is about 240 and our current alternates at 50 Hz. Always remember that our mains are much more dangerous than the American ones. Some tests and expedients that are reasonably safe on American mains may be hazardous on ours.

P. 19: "Alligator clip" is the American term for what we usually call crocodile clips. P. 20: Many American valves (such as the 12AX7 mentioned as the American equivalent of our ECC83) are listed by suppliers in Britain. Equivalents-lists are available which enable you to identify and link up the various respective types. Transistors also have their near-equivalents or comparable types, though they are not always as directly interchangeable as valve equivalents are. There is more individual variation in characteristics and small alterations to associated component-values may be necessary when substituting "comparable" types of semiconductors.

P. 26: phono cartridge is of course the gramo pick-up cartridge; phonograph is the American equivalent of our term gramophone.

In the chapter on Hum Problems, the reference to "line frequency of 60 cycles" refers of course to the American mains AC as already explained. In our case the corresponding figures for this paragraph would be 50, 100 and 150 cycles (or Hz). P. 32: the plug and socket shown in the sketch are the American domestic 2-pin reversible type. Bear in mind that our plugs and outlets are different, and also differently arranged as regards system (earthing and so on). So be careful; and adhere to manufacturer's instructions appertaining to your hi-fi equipment and power supplies. Don't make any alterations that could increase risk of shock etc.

P. 36: Earthing arrangements here can be different from those prevailing on American systems. So don't try direct earthing of equipment without consulting a competent technician or electricity supply authority. Unorthodox alterations can, in some cases, nullify existing safety provisions in the system and cause unnecessary hazards not only in the equipment directly concerned but in other appliances in the household connected to the mains wiring. In the circumstances it would be best to skip the two paragraphs headed "Ground to earth" and the diagram Fig. 406, if you live on this side of the Atlantic!

P. 44: In the paragraph headed "Transformerless components" the warning about "live" chassis hazards applies with redoubled force over here. If the American 117-volt supply can be described as lethal, you can well imagine that our 240-volt mains pack twice the punch! So don't try using transformerless sets in conjunction with any other equipment.

Always take all necessary safety precautions, and in any case of doubt on any point regarding high voltage and similar hazards, seek expert advice from a competent local technician.

P. 55: the mains filter suggested is rated for American 117-volt mains. On our mains, any equipment used has to stand up to about double the voltage of the American mains, so the test voltage and working voltage of components must be correspondingly higher. If you are in any doubt, seek expert advice as already counselled.

P. 118, Fig. 903 relates to the American type of 2-pin mains plug, not of course to ours; so disregard this idea. Voltage references on p. 119 are, again, American as already explained earlier on.

As you will probably have noticed, the American text of this book was originally written some years ago. Much progress has been made since then in the design and production of hi-fi audio equipment. Nowadays, solid-state circuitry and integrated circuit devices are commonly used in most electronic equipment; but hi-fi audio is one of the few fields in which the landslide into transistorization has been more gradual and less total than in some other branches of radio etc. Some enthusiasts still prefer valves for high-quality, high-volume audio amplifiers; but it is rather a matter of opinion. Solid-state techniques have many advantages in certain respects.

If you are to cure troubles you must have something to cure them with! Most servicing entails replacement of at least some components. Whether you should use "exact replacement" parts or general-service substitutions depends largely on the set and the circumstances. You should go into the problem carefully before choosing.

There are several dependable leading firms which stock a vast range of components, accessories, test-gear, tools, and servicing aids or repair-materials by many first-class manufacturers.

Among the various suppliers whose names, addresses, catalogue prices, etc. may be found in the advertisement pages of current technical journals, two typical examples are Home Radio (Components) Ltd. of Mitcham; and Laskys Ltd. of London and elsewhere (they have a number of branches).

In addition to stockists of general components, there are mail order firms etc. which specialize mostly in supply of replacement valves and a vast assortment of solid-state devices comprising hundreds of different types of transistors, FET's, semiconductor diodes, etc. as well as a good selection of modern integrated-circuit devices.

Here again the way to find up-to-date information is to search the advertisement pages of the technical Press.

contents

Introduction

IN ITS EARLY DAYS HIGH FIDELITY WAS A RATHER EXCLUSIVE ACTIVITY, mostly confined to those who were technically sophisticated in audio matters or who exerted themselves to become so. As to what to buy, where to buy it and how to match and adjust components for proper operation, the audiophile was largely on his own. All this has changed. High-fidelity components now are readily available to and can easily be assembled by anyone, regardless of his technical proficiency. Hence millions of such components have found their way into the home.

Nonetheless, high-fidelity equipment continues to be quite sophisticated. The more sophisticated the product, the more likely it is to be subject to trouble. To aggravate the situation, in high-fidelity we are in constant pursuit of perfection. The average owner of a TV set is generally tolerant of round objects that aren't quite round and of straight lines that aren't quite straight. But the typical audiophile tends to go into a dither if his audio system shows the least sign of distress. Accordingly, the definition of trouble as applied to high-fidelity includes, not only complete breakdowns, but also all the minor deviations from the very best performance of which the equipment is capable — the kind of performance demonstrated in an audio salon or at an audio show.

Therefore we must appreciate that superior sound is not the sole distinction between high-fidelity and ordinary equipment. Also important is freedom from trouble. But everything is relative, and we can only say hopefully that we expect high-fidelity components to be more troublefree than the garden variety. Considering the complexity of high-fidelity equipment and that the owner tends to be unforgiving of even the most minor aberration, it would be a mistake to assume that the purchase of such equipment automatically provides immunity from trouble.

This is not meant to dim the luster of high-fidelity. It is simply being realistic. You hope that everything will work correctly from the start. And you hope that the system will continue that way for an indefinitely long period. But you must be sensible and prepared in case things turn out otherwise.

Then what do you do if a component is inoperative or operating below par? Naturally, the first thought is to go to a technician (perhaps the dealer who sold the equipment). But the fees of a competent audio technician, although deserved, are not low. Moreover, if the equipment is working but not quite to one's satisfaction, it may be awkward to prove and pursue the complaint.

Fortunately, a technician is not always needed. The audiophile can do many things to discover the cause of his complaint and remedy it. The purpose of this book is to discuss the various types of audio troubles, their causes and the extent to which the audiophile can remedy them. It is my hope that through the suggestions appearing here, the reader will get the maximum pleasure from his high-fidelity equipment.

Some materials in this book have been taken from articles written by the author, and he wishes to thank the editors of *Audio, HiFi/Stereo Review* and Radio-Electronics Magazine for permission to use them. Thanks are due to *High Fidelity* magazine for permission to borrow two drawings (Figs. 601 and 602) from the February 1963 issue illustrating how a living room might be furnished in a "hard" or "soft" manner; and also to Acoustic Research, Inc. and Electronic Instrument Co. (EICO) for illustrations supplied by them.

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Chapter 1

Why We Have Audio Troubles

FROM TIME TO TIME INDIVIDUALS AND ORGANIZATIONS THAT TEST audio equipment report that something like 25% to 33% of audio components as they come out of the carton contain a defect, although usually a minor one. To be on the safe side, we may take the lower figure and slice off a chunk, leaving 20% as a conservative estimate of the percentage of components having a defect, whether minor or major. The significance of this figure must be examined in light of the number of components that ordinarily go into an audio installation. Following is a typical list:

Phonograph (turntable, arm and cartridge)

Tuner (and perhaps a separate multiplex adapter)

Preamplifier

Power amplifier — or combination preamp-power amplifier or even a combination tuner-preamplifier-power amplifier (called a receiver).

Speaker (s)

Assume you have an audio system consisting of five individual components (Fig. 101). What is the probability that these can be taken out of their cartons, be assembled without benefit of testing, and work perfectly right from the start? The answer is 33%. There is about one chance in three that your audio system will work correctly in every way when you first assemble it.

This shouldn't be too surprising. Considering that a stereo system may contain several dozen tubes (or transistors), several hundred resistors and capacitors, and numerous other electrical and mechanical parts, there is much that can go wrong. Quality control by the manufacturer is, of course, the answer. But quality control is expensive and the audio market is highly competitive and price-conscious. It is difficult for a manufacturer to exercise as rigorous control over a consumer component selling for $150 as over a similar professional piece of studio equipment selling for $500 or more.

An audio system may easily contain more than five components. It may include a tape recorder, a multiplex adapter (to convert a mono FM tuner to stereo), one or more electronic crossovers (for channeling the low frequencies to one power amplifier and the

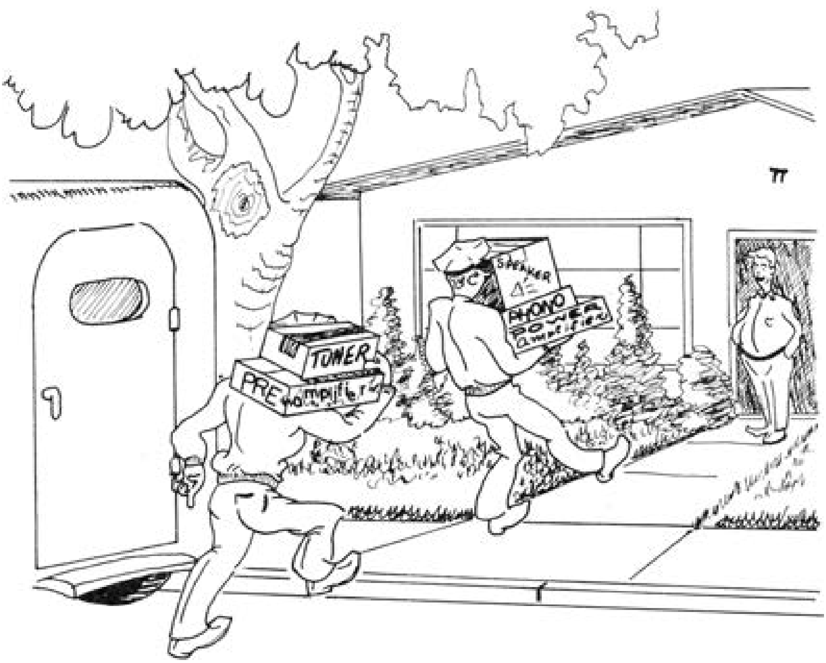


Fig. 101. *Typical components of a hi-fi system.*

high frequencies to another), an electronic reverberation device, a dynamic range expander and so forth. To the extent that there are more than five individual components, the chance of immediate success is reduced well below 33%.

From this we draw an important conclusion: If at all possible, have each component checked before you buy it (Fig. 102). If you buy all the components at one time, buy them at one place and have them checked, not only individually, but also as an ensemble (Fig. 103). This points to the importance of dealing with a reputable audio store, one willing to spend time and effort in checking

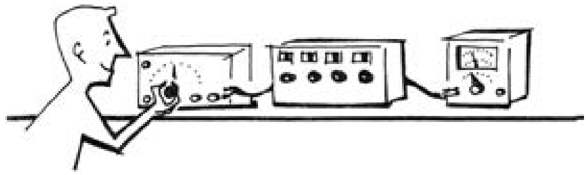


Fig. 102. *Technician checking an audio component.*

components. And it points to the importance of making your purchase in person, so that you can witness the final checkout of the components.

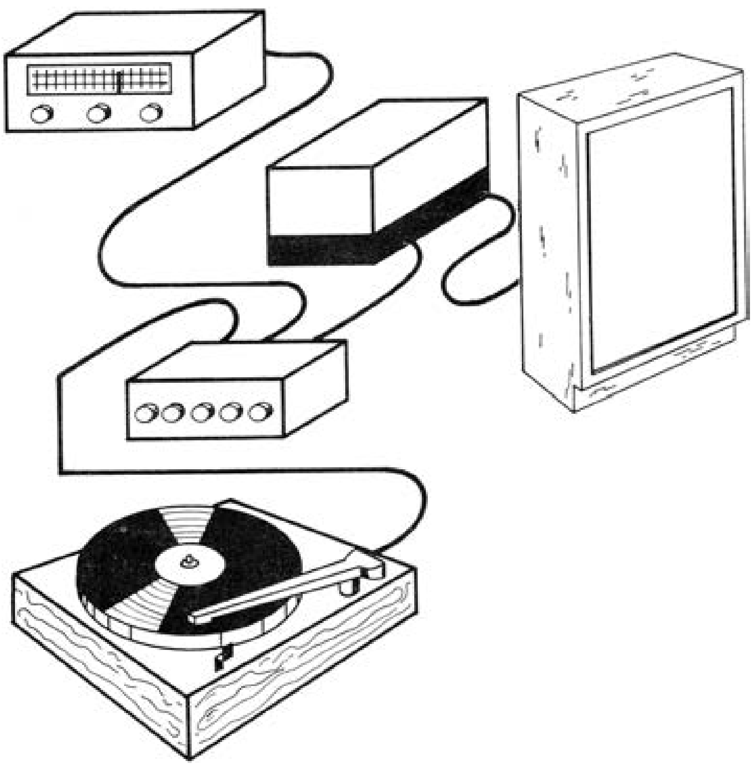


Fig. 103. *Audio system assembled for an ear check. While separate components are shown here, many systems consist of integrated components, such as a preamp and power amplifier on one chassis, or a preamp, power amp and a tuner, all on one chassis.*

Pretesting of components by the dealer just can't be overstressed. It is almost as important as the brand of component you choose to buy. However, some salesmen, when asked to check a component before you take it home, reply, "That isn't necessary. It comes in a factory-sealed carton." Since you are not the least bit interested in the condition of the carton but of its contents, you should promptly place distance between yourself and this type of salesman.

On the other hand, perhaps you can't really blame the salesman altogether. Perhaps you are partly or even completely to blame. Quite possibly, after making the rounds of a number of audio stores, you have beat a salesman down so far in the matter of price that he just can't afford to do anything more than wrap your package and take your money. Checking a component takes time. Since time is money, he may not be able to afford to check your components at the price you have wheedled from him. For example, consider a stereo preamplifier with its number and variety of knobs, switches and input and output jacks. It is not a small thing to put such a component through its paces to make sure that everything works just right.

Yet this is surely what you want. The best insurance you can buy against audio troubles is a complete check of a component before you pay for it and take it home. Not a superficial check; for example, you don't want to find out six months afterward that one of the input jacks of your preamplifier doesn't work. An adequate check is apt to cost you something, perhaps adding as much as 10% of the purchase price, but it is well worth it. There are many cases where an individual arrived home enthusiastic about a newly acquired group of components and expectantly hooked them together, only to have his enthusiasm sour because something or other didn't work properly. The unnecessary delays and expenditure of energy are frustrating.

Thus, the very first step in warding off audio troubles is to make sure that what you buy has been checked out thoroughly by competent personnel and demonstrated in your presence to be working properly.

Given an audio system that works properly, you have reason to expect it will remain that way for a period but not forever. All components (except possibly the speaker) inevitably run into trouble, just like your refrigerator, washing machine, automobile, TV set, etc. How long will it be before you can expect trouble? There is no way to say for sure. It may be a year or two or even three before something goes wrong. Or it may be only a few months.

To some extent, this is a matter of luck. Discounting luck, we can say that, as a general rule, the frequency of trouble depends upon the quality of the parts employed in the audio component and upon how much use you make of the component.

Thus, we can reach a second conclusion. The more you expect to use your audio system, the more money you should put into it in terms of getting high-quality components. The difference between expensive and moderate-cost components is often imperceptible to the ear; it is quite possible to find an amplifier costing around $150 that sounds hardly different than one costing around $250 (Fig. 104). The principal difference may well be in the

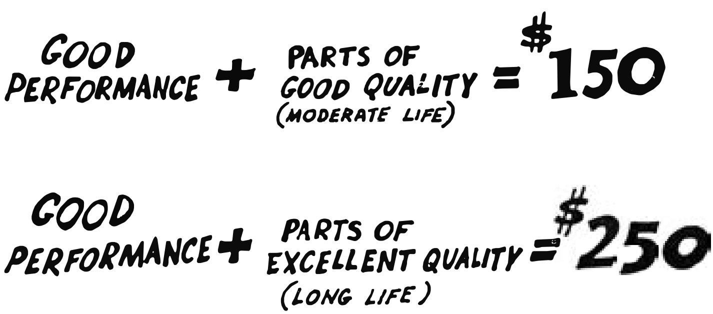


Fig. 104. *Why some components cost more.*

quality of the parts that go into each, assuring longer life and greater freedom from trouble for the more expensive unit. Accordingly, if you expect to operate your audio system a great deal and for many hours at a time, it is probably least expensive in the long run to buy components of top quality. On the other hand, if you expect to listen to your system only a few hours each week, a suitable course may be to get components of moderate cost, provided that there is no audible reduction in sound quality. However, when higher price corresponds to superior performance—such as the ability of an FM tuner to bring in distant stations with low noise and distortion—the degree of use tends to become a minor consideration.

Seldom do tubes, resistors, capacitors and other audio parts go bad all of a sudden. More often, they deteriorate gradually. Accordingly, they may introduce gradual deterioration into the performance of a component, perhaps so imperceptibly that you are a long time in noticing that something doesn't sound right. What is worse, the slow deterioration of certain parts may endanger other parts, possibly ones quite expensive to replace. For example, a faulty filter capacitor may place an undue strain on the power transformer; whose replacement can cost you a pretty penny. How are you to know whether deterioration is setting in? Well, often you really can't know. But you can take preventive measures, on the same principle that you should see your dentist and doctor periodically for a checkup.

So we arrive at a third conclusion: Part of the process of thwarting audio troubles is to have your equipment checked periodically by a *competent* technician. If feasible, have this done once a year, but not less often than once every two years (Fig. 105). Keep in

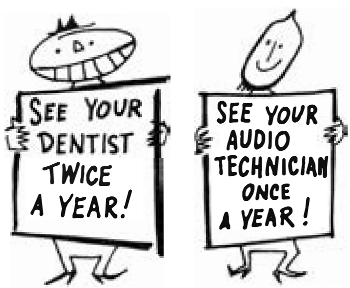


Fig. 105. *A stitch in time . . .*

mind that there is a vital difference between your high-fidelity equipment and other audio equipment, such as your kitchen radio. Generally, you don't bother to do anything about your kitchen radio unless it quits completely or sounds utterly bad. Not so for high-fidelity equipment. From this you expect not merely performance but performance-plus. You expect the equipment to be operating in top-notch fashion, because therein lies the difference between high-fidelity and other audio equipment.

Reviewing the three conclusions we have reached, if you want to minimize the chances of having trouble with your audio equipment:

1. Have the equipment thoroughly checked before you pay for it.
2. Buy equipment of better than average quality if you plan to put it to considerable use.
3. Have the equipment checked periodically.

When you eventually encounter troubles, these will fall into two categories:

1. Those you can cure yourself.

2. Those that require an audio technician.

Although the dividing line between these two categories is not very sharp—some audiophiles have a moderate amount of technical competence acquired through building kits or in other ways—there is a dividing line. A variety of troubles can be cured only by the person with a substantial amount of technical knowledge and an array of technical instruments. Hence the role and importance of the audio technician is not to be minimized. On the other hand, many elementary problems can adequately be taken care of by yourself.

A parallel may be drawn between first aid and the services of a doctor (Fig. 106). You don't have to go to a doctor for every

Fig. 106. *First aid may be enough.*

ache, bruise, cut and illness that besets you. Nor does every malfunction of your audio system require a service technician. The simple things which you can do are often the same as the first steps that a technician might take in approaching a problem. Then, if you have failed to cure the problem, is the time for the technician to enter the picture. At the same time, it is important to know what you can't do, because unwise tinkering may lead to extra expense.

In the main, this book is concerned with troubles of an electronic rather than mechanical sort. Mechanical troubles, such as may afflict a phonograph or tape recorder, tend to be more individualistic than electronic difficulties as you go from one brand of component to another, and their repair almost inevitably requires special skill, knowledge and parts.

This book doesn't purport to cover every type of audio component, but limits itself to the basic ones: speaker, power amplifier, preamplifier, tuner, phonograph and tape recorder. There are sufficient similarities between these and other varieties of equipment so that discussion of the basic components will also apply substantially to the others.

Rather than devote a chapter to each basic component, it appears more useful and less repetitious to organize this book in terms of the fundamental problems that generally occur:

- Hum
- Noise
  - Hiss
  - Other (crackles, clicks, pops, frying sounds, etc.)
- Poor Bass
  - Not enough
  - Too much
- Poor Treble
  - Not enough
  - Too much
- Distortion
- Insufficient Sound
  - None at all
  - Weak
  - Intermittent
- Special problems of the stereo installation
- Problems of kit building

Even before getting into these problems, we shall discuss some of the simple tools and aids (costing little or nothing) that may be helpful in rooting out audio troubles. And we shall discuss the art of discovering in what component or in what section of a component the trouble lies.

Chapter 2

Tools of the Trade

IF YOU PEEK INTO THE BACK ROOM OF A WELL RUN RADIO–TV REPAIR shop, you will generally find an impressive array of equipment, including an audio oscillator, radio-frequency generator, vacuum-tube voltmeter, other meters, oscilloscope, capacitance checker, distortion meter, soldering iron or gun and other mechanical equipment, manuals containing electronic and mechanical diagrams, spare parts, etc (Fig. 201). All these and even more are

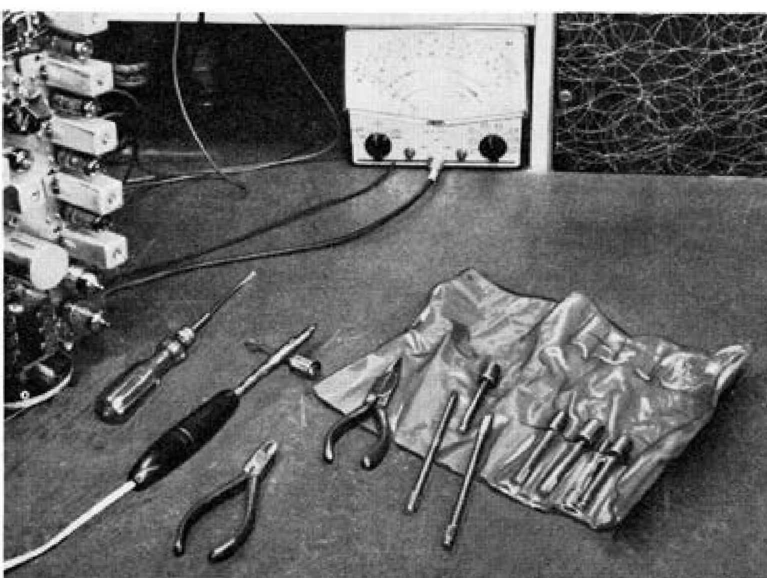


Fig. 201. *Service technician's equipment.*

also necessary or desirable for servicing audio equipment. These are the tools of the technician's trade.

Does this mean that you have to acquire a somewhat similar assemblage of equipment before you can cope with the problems that may crop up in your audio system? By no means. There are two reasons why we are calling your attention to the equipment used by the professional: First, to alert you to the fact that you, too, but in a far simpler way, must equip yourself to handle audio problems. Second, to remind you that you won't be able to solve every difficulty that comes along, because a number of these difficulties do require the professional instruments and know-how of the competent technician.

What do we mean by *your* tools of the trade? Suppose we list them, and then explore them in detail:

1. A signal source (costing a few cents or nothing at all).
2. A device for checking electrical continuity (costing a few cents).
3. Two or more cables for making connections from one point to another (costing a few cents each).
4. Spare parts (mainly tubes, costing more than a few cents each but economical in the long run).
5. Instruction and service manuals (which generally come with the audio components, costing you nothing to *keep* them; if you have to buy one, the price is usually but a dollar or two).

Signal source

One troubleshooting technique used by the audio technician is to inject a signal (audio tone) into the component being investigated, so that a sound may ultimately come out of the speaker. To make suitable adjustments or repairs, he may need to know if *any* sound comes out or what *kind* of sound comes out. The professional employs a variety of signals for various purposes, but there is no need to elaborate on this. The important thing is to alert you to the fact that sometimes you will want to introduce a signal into your audio installation without relying on your tuner, phonograph or tape recorder.

Every man is his own signal source when he is in a building that provides 60-cycle current, as is generally the case. The alternating current that runs through the walls of your home, apartment house or other building produces a magnetic field which, though slight, is sufficient to be picked up by your body. As a result, if you touch something (Fig. 202) you are applying a small amount of 60-cycle voltage to it. You can easily prove this by touching the leads of a

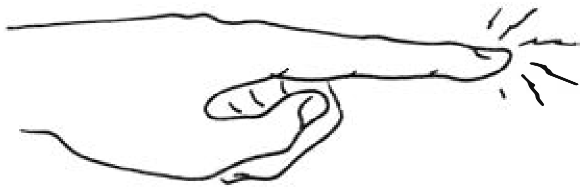


Fig. 202. *One's self as a signal source. See also Fig. 605 on page 69.*

very sensitive voltmeter and noting how the pointer swings.

How might you use this faculty of yours? Suppose that you doubt whether your phono cartridge is working properly, because the signal appears very weak when you turn the selector switch of your preamplifier to the phono position. Or the preamp might not be functioning properly. Therefore you bring your finger near the leads going to the cartridge terminals. If the volume control of your audio system is turned up moderately, the speaker should emit a substantial 60-cycle note. If it doesn't, this suggests that something is wrong with your preamp rather than with the cartridge. But if your audio system responds with a healthy roar—don't let it roar too loud lest the speaker be damaged—the blame shifts to the cartridge.

As another example, assume you want to check the phasing of a pair of stereo speakers (that is, to check whether the cone of one speaker moves outward when the cone of the other speaker also moves outward, instead of the two cones moving in opposite directions). There are several ways of making this test, some of which employ a low-frequency tone such as might be produced by an audio oscillator, test record or test tape. But if you have none of

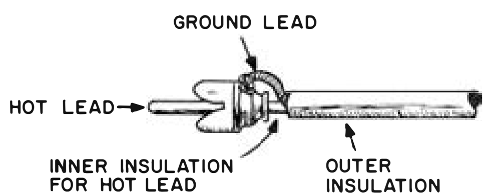


Fig. 203. *Shielded audio cable. See also Fig. 407 on page 37.*

these, you still have yourself as a signal source for phasing. Simply take one of the cables leading into the amplifier, for example, the cable from the tuner to the amplifier, disconnect the end that goes to the tuner, and touch your finger to the "hot" lead—the inner portion of this cable—as shown in Fig. 203. You should get ample 60-cycle sound for phasing purposes.

To take one more example, you may want to verify whether all the input jacks of your amplifier are functioning, that is, capable of passing a signal to the selector switch. To check any input jack, you merely have to plug in a shielded cable, touch your finger to the hot lead, and turn the selector switch of the amplifier to the position corresponding to that input jack. If you hear a 60-cycle note coming clearly and fully through the speaker, the input jack in question is operative.

Earlier in this chapter we mentioned that the signal source might cost a little something. This little something is 15 or 20 cents for a battery. There are situations, particularly in testing a speaker, where the result of briefly connecting a battery can tell you something you want to know. But more of this in another chapter.

Continuity checker

There are a variety of situations where you may want to check whether a lead or cable is intact and capable of passing an electrical signal. Such a lead or cable may be too long to check by eye, or part of it may be hidden from view. A continuity checker enables you to discover whether the lead has a break in it.

To make a continuity checker, all you need are a flashlight battery, a flashlight bulb and several feet of wire. Fig. 204 shows how

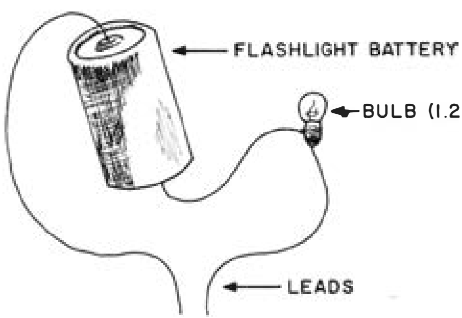


Fig. 204. *A simple but effective continuity checker.*

these items are connected. Soldered connections are best but, if you don't have access to a soldering iron, surely you can figure out a mechanical way of holding the items together, even by such primitive means as cellophane tape. Fig. 204 shows that one terminal of the battery goes to one terminal of the bulb, while the other terminal of the battery and of the bulb are each connected to several feet of wire, which form the leads of the continuity checker. The bulb lights when you touch these leads together, because you have completed the circuit. This is the basic principle of the continuity checker. If you are checking a cable (Fig. 205)

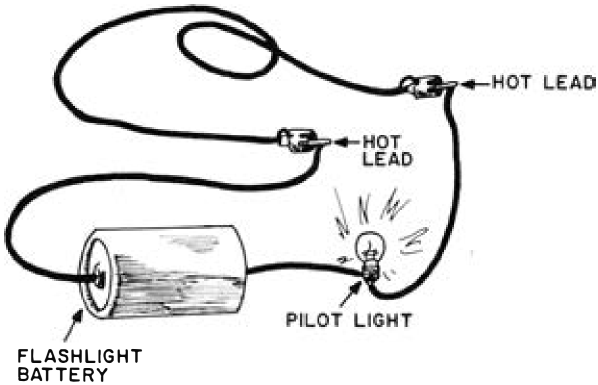


Fig. 205. *Testing a cable with the continuity checker.*

and apply the checker's leads to opposite ends of the cable, the bulb lights only if continuity exists. Failure to light indicates a break. Intermittent lighting (for example, the bulb flashes on and off as you wiggle the cable being tested, indicates an intermittent connection) .

If you happen to own or have access to a vom (volt-ohm-milliammeter) or vtvm (vacuum-tube voltmeter), these can be used to check continuity. Such instruments read resistance. Zero or very close to zero resistance denotes a continuous electrical path. If you set the meter to read resistance and apply its leads across the ends of a cable, zero reading indicates continuity.

Connecting cables (patch cords)

There are times when you will want to make a quick, temporary connection between two points. For example, you might want to place a temporary short circuit across one end of the leads to your FM antenna to check continuity. Or you might want to make a temporary, readily detachable connection between a speaker and a power amplifier.

For these and similar purposes, it is desirable to have at least two cables, each about 2 or 3 feet long, with an alligator clip at each end (Fig. 206) . An alligator clip costs about 10 cents, and flexible "test-lead cable" costs about 5 cents a foot. Either you can solder the test lead to the alligator clip, or you can get the type of clip where the test lead is fastened by a screw.

It is advisable to use varying colors for your connecting cables, so that you can readily distinguish among them when using more than one at a time. In case you don't use varying colors, use dabs of fingernail polish at each end of a cable to provide identifying marks.

Spare parts

By spare parts we mean only items which can be plugged in and not those which have to be wired in. This would include components such as tubes, transistors and fuses. Occasionally, one finds other plug-in parts, such as a filter capacitor. (Not all transistors are plug-in types. You will find some soldered into position.)

For the most part, we are concerned with tubes (or transistors). If you consult the instruction or service manuals accompanying your audio units that contain tubes, you will probably find a listing of the tubes employed and a diagram showing the physical location of each (Fig. 207). You can almost always get manuals by writing to the manufacturer. If you can't get a tube list and location diagram, you can remove the top cover (if any) from the unit in question and ascertain by eye what tubes are used and where. In a few moments, you can make your own tube-location diagram.

You should have on hand a replacement for every tube in your audio equipment. Considering that your equipment may contain two or three dozen tubes and that the cost may average about $1.50 to $2.00 per tube (at the price you ordinarily pay at electronic supply stores or mail-order houses), this sounds like a substantial investment. However, you will find that the same type of tube is used more than once in most components. Thus, in a stereo preamplifier, all six or seven tubes may be of just one or two types. Accordingly, your investment is considerably reduced.

Moreover, you will probably find that the same tube type is used repeatedly in several components. For example, you may find 12AX7's in your tuner, preamplifier, power amplifier and tape recorder. Thus your investment is further reduced.

You may even find that what appear to be different tube types are really the same except for different designations. For example, the 12AX7 is also known as an ECC83, the latter being the European designation for what is electrically the same tube. They are completely substitutable, except that the European version tends to have better characteristics with respect to hum and noise. If you have an ECC83, you can use it as a replacement for a 12AX7, although you can't always do it the other way around. The United

States also makes a superior version of the 12AX7, known as the 7025. You can probably use this quite satisfactorily to replace either

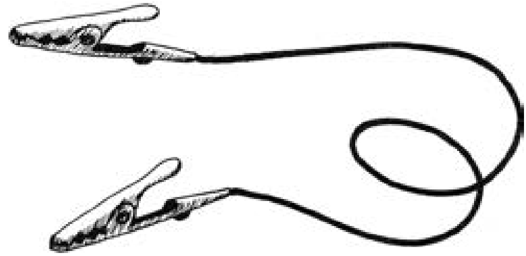


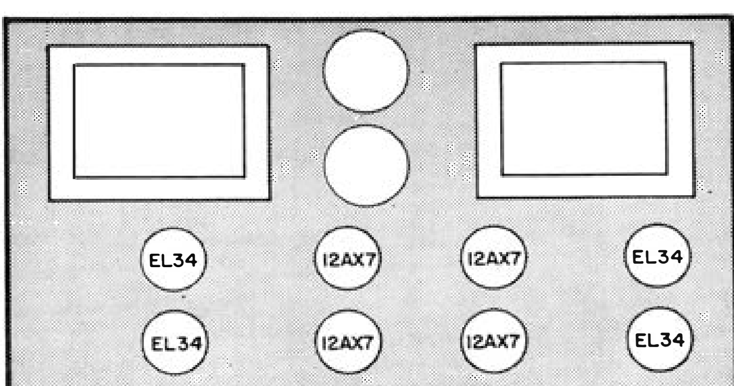
Fig. 206. *Connecting cable with alligator clips.*

a 12AX7 or ECC83. Similarly, there are other interchangeable tube types.

After you have made a listing of each tube type for your replacement stock, consult your audio dealer as to which of these types are equivalents, enabling you to pare the list of replacement tubes to a minimum. In the final analysis, you may well find that you need only a relative handful of replacement tubes (Fig. 208) representing a modest investment compared with the cost of your audio system and the worth of a prompt repair.

It is quite possible that you may never get the opportunity to use some of the tubes you have acquired as spares. But weigh this possibility of an unnecessary expenditure, probably involving less than $10, against the inconvenience and delay in going out to buy a replacement tube, or against the cost of a single visit to an audio repair shop. Finally, consider that you will have made this unnecessary expenditure only once, compared with listening

Fig. 207. *Tube-location diagram.*



satisfaction over a number of years. You can draw a parallel with fire insurance—you are better off if you never collect it.

Generally, it is sufficient to have just one replacement tube of each type. However there is an important exception to this statement. The exception is the output tubes of a power ampli-

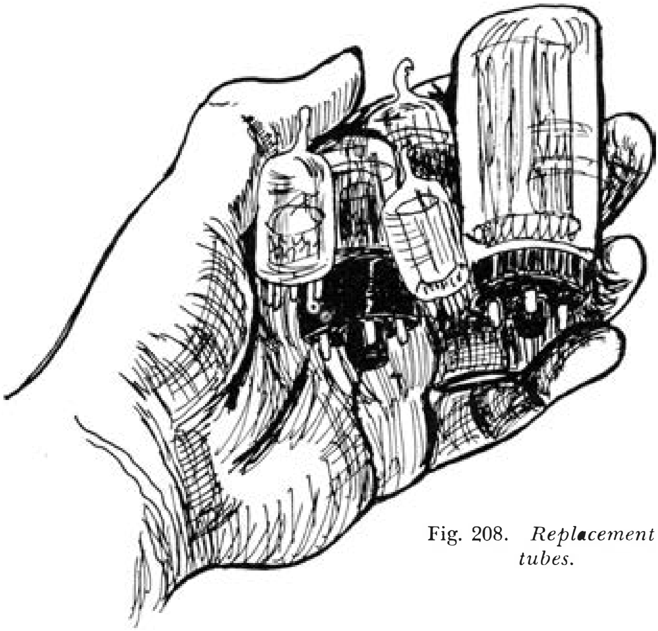


Fig. 208. *Replacement tubes.*

fier. It is standard practice in high-fidelity power amplifiers to employ a *pair* of output tubes in what is known as a push-pull circuit. In the case of a stereo amplifier, two pairs of output tubes are used. Your replacement stock of output tubes should always consist of pairs; a "matched" pair (Fig. 209), which commands a moderate premium in price, is preferable.

Even among tubes of the same type, there are slight differences in performance from one to another. These are usually not very important; still, they may be of some significance in high fidelity. A matched pair of tubes consists of two units, selected by the manufacturer, having nearly identical performance characteristics. By using a matched pair of output tubes you make it possible for the power amplifier to yield the best performance that its design permits.

If one output tube requires replacement but the other still appears to be in good condition, it is nevertheless wise to replace both at the same time. The output tubes are the hardest workers of all, subject to continual degradation of performance. When one tube has reached retirement age, the other is apt to be not far behind, so that it doesn't pay to coax a little more service

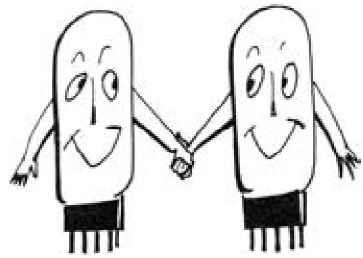


Fig. 209. *Output tubes go in pairs.*

from it. Even if the still-good tube has many months of life left, it is not likely to make a good match with the new tube freshly put in.

Manuals

Audio components are almost always accompanied by an instruction manual; sometimes by a service manual as well. No matter how familiar you think you have become with the operation of your system, it is wise to keep the instruction manual in a well remembered place where you have ready access to it. And no matter how little use you may think you have for the service manual, because you understand little or none of it, it pays to keep this handy too.

If you have just purchased a group of audio components, it takes considerable will power to curb your impatience to hear how they sound in your home. The temptation is great to assemble them immediately, giving the instruction manuals a cursory glance if any. But by doing so you risk getting less than maximum performance from your system. There is even some risk of doing damage to a component. For example, you might connect a ceramic pickup to the magnetic phono input jack of the preamplifier, resulting in a blast of sound possibly great enough to harm the speaker. Or you might not connect the speaker properly to the output terminals of the power amplifier, so that there is no load across the amplifier and hence a possibility of damage to the amplifier. Or you might turn the wrong control on the power amplifier in the belief that you are reducing its input level set, but actually you might be changing the bias on the output tubes and hastening the end of their lives. And so forth.

By all means read the instructions first (Fig. 210). After you have done so and your system is operating satisfactorily and you are thoroughly familiar with its operation, even then don't throw away the instructions. If trouble arises at a later time, a rereading of the instructions may provide a clue as to where the difficulty lies. The instructions may contain specific troubleshooting hints; in the case of kits, they usually do. The instructions may contain a circuit

Fig. 210. *Always read the instruction manual first.*

diagram which may baffle you but can be of considerable aid to the audio technician. True, the technician should have his own diagram, but if he doesn't and you supply him with one (make sure you get it back), this can facilitate prompt service and perhaps reduce the cost of that service.

For much the same reasons it pays to keep the service manual, if you have one. Moreover, it is possible that, as time goes by and you learn something about the workings of audio equipment, some of the contents of the manual may become meaningful enough to assist you in the maintenance of your equipment and in making minor adjustments and repairs.

Instruction or service manuals contain charts showing how much voltage and how much resistance should normally be measured at each pin of a tube socket. As the result of building kits, many audiophiles have acquired enough familiarity with components and enough confidence so that, given a suitable meter, they are able to check voltages and resistances. If any of these values depart substantially from normal amounts, this suggests that the trouble is in the immediate vicinity of the tube where the incorrect voltage or resistance is found. A few minutes spent in checking the resistors and capacitors in this area often will uncover the part responsible for the difficulty. Here we are talking about the "advanced" audiophile, but through kit-building and other electronic ventures, thousands of audiophiles have reached or are in the process of reaching an "advanced" stage.

Chapter 3

The Art of Substitution

THE KNACK OF DISCOVERING WHAT IS WRONG WITH YOUR AUDIO system is often the knack of discovering what is *not* wrong. The technique of substitution enables you to put your finger on the trouble in the minimum number of moves. It establishes which parts of your audio system are working correctly, so that whatever is left is the logical object of suspicion.

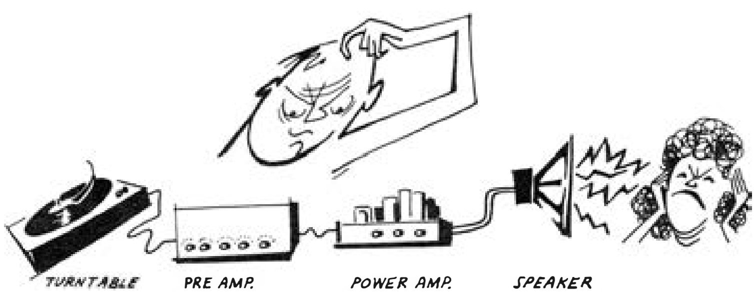


Fig. 301. *Where's the trouble?*

Suppose that you get distorted sound when playing a phonograph record (Fig. 301) . Tubes being the most likely single cause of trouble, your first thought may be to check them. If you have a stock of replacement tubes, you can replace one at a time until the system works properly—assuming the culprit is indeed a tube.

But where do you start? Do you simply replace tubes at random?

Hardly likely, because there are quite a few tubes in your audio system, and some may not be easy to get at. In fact, some or all the components may require, not only that you remove them from a cabinet or other resting place, but also that you remove their top cover in order to have access to the tubes.

On second thought, you may decide not to be so quick about checking tubes, at random or otherwise. After all, possibly the phono cartridge is at fault. Therefore the first step you could take is to listen to your system when the tuner instead of the phonograph is the signal source. If the system then sounds fine, this *suggests* that the cartridge is at fault. But not necessarily. It is also possible that the fault lies in the magnetic phono section of your preamplifier.

At this point, you may decide it is worth extracting and checking the tube (or tubes) in the phono section of the preamp. Or, because of the effort involved in doing so, you may decide it is still too soon to question the tube. Therefore you want an alternative source to feed into the magnetic phono input. Perhaps you own another pickup, the predecessor of the better one that you now own or a leftover from the days when your system was mono instead of stereo (in which case, test the cartridge only on a mono record). If, using a substitute pickup, you get the same distortion as before, it is almost certain that, if any tube is at fault, it is the one in the phono portion of the preamplifier.

Suppose that you don't have access to a substitute pickup. Then you might look to your tuner (Fig. 302) as an alternative signal source. However, your tuner puts out a much greater signal than the magnetic phono input is ordinarily able to accommodate. Here is another source of distortion. Therefore you may connect the tuner to the magnetic phono input jack only if it is possible to reduce the tuner's signal to a level low enough to be comparable with that of a magnetic cartridge. Fortunately, most tuners have a level control that enables you to do this. Then you can connect the shielded cable from the tuner to the magnetic phono input. If the sound appears free from distortion—except for being bass-heavy and treble-shy due to phono equalization—you have indicated the cartridge as the offender. But if once again you get distortion, then it is likely the phono tube after all.

What if you have neither a second cartridge nor a tuner as an alternative signal source? Perhaps you have a tape recorder that can be connected to your audio system or perhaps you have a friend able and willing to lend you one of these.

But suppose that you have absolutely no alternative signal source. Well, things still aren't really so bad. After all, you have narrowed the trouble to either the cartridge or the phono tube, and you might as well proceed to check the tube.

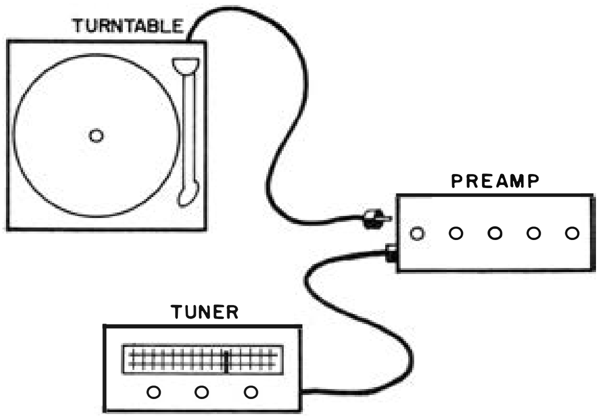


Fig. 302. *Substituting the tuner for the phono cartridge as a signal source.*

Let's change the example somewhat. Suppose that distortion is still the symptom, and that when you operate the tuner (through the proper input jack) the distortion continues. Then the trouble may lie in the preamplifier, the power amplifier or the speaker.

If it is in the preamplifier, you can quickly discover this by connecting the tuner directly to the power amplifier (Fig. 303).

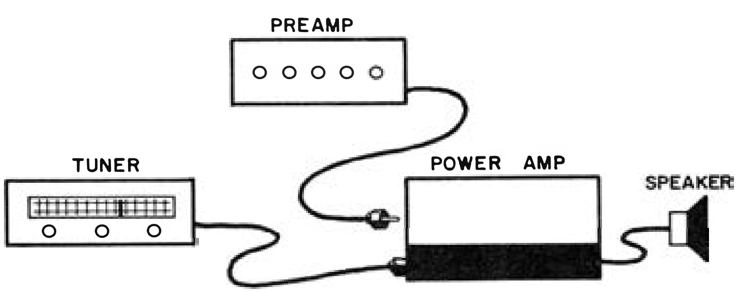


Fig. 303. *Bypassing the preamplifier.*

If the sound clears up when you bypass the preamplifier, then the preamp is at fault. Remember, when you connect the tuner directly to the power amplifier, you must reduce the tuner's volume control, or else the amplifier's input level set, to avoid the possibility of blasting the speaker.

If the distortion still exists when you bypass the preamp, the trouble lies in either the power amplifier or the speaker. Try connecting the power amplifier to another speaker. Perhaps you have an auxiliary speaker in another room of the house that you can borrow for the present purpose. If your system is stereo, borrow the speaker from the untroubled channel. As a last resort, you might use the speaker in a table radio. Disconnect the radio from the power line, open the case and locate the speaker terminals. Without breaking any leads in the radio, connect your amplifier to the radio speaker by connecting cables (Fig. 304).

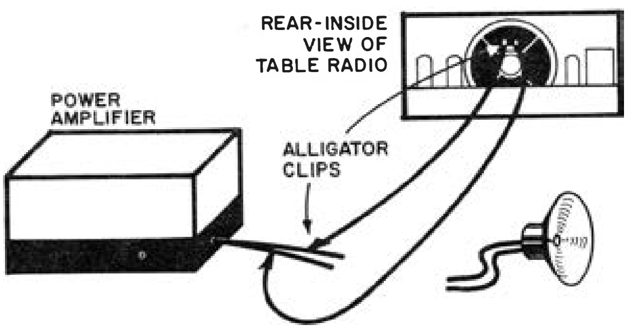


Fig. 304. *Using the speaker in a table radio as a substitute.*

If the sound is as clear as you can expect from such a speaker, this indicates that your high-fidelity speaker has developed trouble. But if the sound remains distorted, the fault is in the power amplifier.

Instead of using a substitute speaker, it is possible to use a substitute amplifier. You might have a second high-fidelity system in the house, or you may have a mono amplifier left over from your pre-stereo days. Or you might have an old radio–phonograph console that has been relegated to the den or playroom. In each case it may be feasible to test your speaker by connecting it to the substitute amplifier.

Stereo, which requires two of most components, offers extra opportunities for substituting one component for another in tracking down malfunctions. If you suspect a speaker, you can substitute one stereo speaker for the other; it is quite unlikely that both would go bad at once. Similarly, you have two channels in your preamplifier and two in your power amplifier—more opportunities for substitution. (On the other hand, there is a certain amount of electrical interdependence which makes it possible for both channels occasionally to go bad at once.)

Assume that the sound is poor only on the left channel when playing the phonograph. Connect the left-channel cable from the phonograph to the right phono input of the preamplifier. If the sound is good on the right channel, this absolves the cartridge. But if the sound remains poor, this indicates that the left section of the cartridge is faulty.

Assume that the sound is good when the left section of the cartridge is fed into the right channel of the preamp. Then you are faced with the task of finding out whether the trouble lies in the left channel of the preamp, the power amplifier or speaker system. We have already discussed how easy it is to discover whether the speaker is responsible merely by substituting one speaker for the

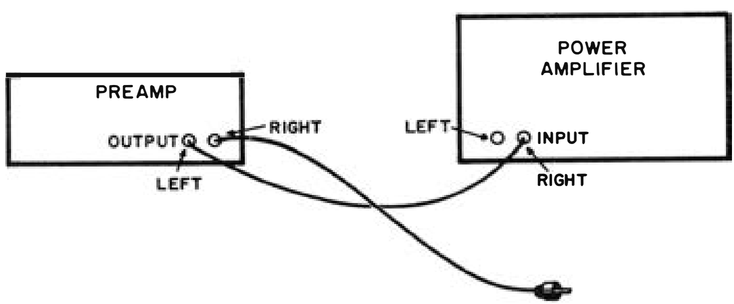


Fig. 305. *Checking whether trouble lies in the preamplifier or power amplifier.*

other. If the sound continues to be poor on the left channel, the search narrows to the preamplifier or power amplifier. Now connect the left channel of the preamp to the right channel of the power amplifier as in Fig. 305. If this clears up the sound, the left channel of the power amplifier is defective. If the sound is still faulty, the left channel of the preamp is defective.

In attacking any problem don't overlook the possibility that the trouble lies in one of the interconnecting cables between components. For example, a complete or intermittent short in a cable would cut off the sound completely or do it spasmodically. Therefore in locating troubles by the substitution method, note what happens when you introduce a different shielded cable between a pair of components, for example between the tuner and the preamp or between the preamp and power amplifier. Also check out what happens when you twist or bend a cable, because this too may reveal an intermittent short or open condition.

When you have narrowed your search for trouble to a particular component, it may turn out that it is outside your capacity to solve the problem. This does not mean your search has been in vain. You will still save time and money, because now you will have to bring only one component instead of several to the audio technician. If you had no idea whatsoever of where the trouble lay, you might have decided to have the technician call at your home—a still more expensive matter.

Chapter 4
Hum Problems

HUM IS INEVITABLY PRESENT IN EVERY AUDIO SYSTEM THAT OPERATES from alternating current. The line frequency of 60 cycles is the principal offender, but harmonics (multiples) of the line frequency, particularly 120 and 180 cycles, also threaten listening pleasure. Hum may be moderate, constituting only a mild annoyance. Or it may come with a roar that endangers components, particularly the speaker (Fig. 401).

One objective of high-fidelity equipment is to keep hum so low that it is virtually inaudible. With top-quality components, you

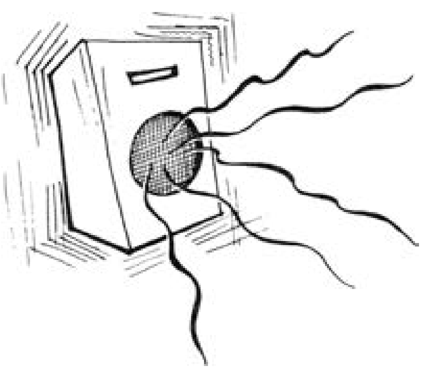


Fig. 401. *Violent hum can damage a speaker.*

will hear extremely little if any hum at more than 2 or 3 feet from the speaker. With mediocre equipment, a relentless undercurrent of hum may prevade the room, becoming especially noticeable at a quiet hour and during soft passages of music.

It is outside the scope of this book to suggest ways of reducing hum through the modification of equipment. We must assume that you have reasonably well designed equipment to start with.

But hum may rise to offensive proportions after a period of use or because you haven't assembled and adjusted your components correctly. The trouble may require a technician; for example, if a filter capacitor has gone bad. But in many instances it may be within your ability to forestall or reduce hum.

Which component has the hum?

The first step is to discover which component is responsible for the hum. Refer at this point to Chapter 3 on the art of substitution. However, let us consider just one example here. Assume that you hear hum when the tuner is on, but not when the phonograph or tape machine is played. Then the finger of suspicion points at the tuner. However, if you hear hum no matter whether the tuner, phonograph or tape machine is played, the preamplifier or power amplifier is at fault. Disconnect the preamplifier from the power amplifier. If hum remains, the trouble is in the power amplifier; if hum disappears, the preamplifier is at fault.

Plug reversal

An elementary yet sometimes effective method of reducing hum is to reverse the position of the line plug in the 117-volt socket (Fig. 402). Try this for each component, starting with the preamplifier and power amplifier (or integrated amplifier). Any hum reduction that occurs will be moderate but not to be overlooked

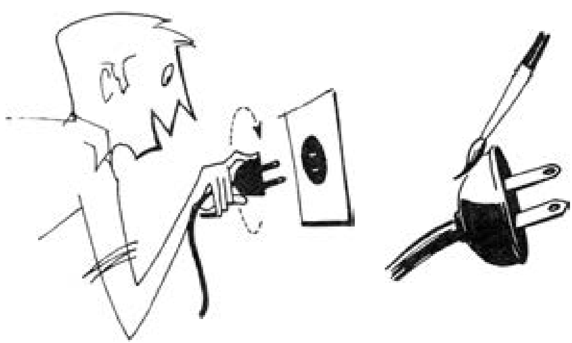


Fig. 402. *Reversing the line plug.*

—a bit of improvement here and a bit there can add up to a significant total. If you find that the plug position affects hum, identify the best position with a dab of fingernail polish on the plug and socket.

In the case of the turntable, try plug reversal with the motor running. Similarly, test the tape recorder with its motor running. Don't simultaneously operate components that ordinarily are not used at the same time, for example, the tuner and the turntable.

When testing for least hum, keep in mind that hum intensity will vary at different points in the room. Therefore choose a listening spot where hum is loudest. You may need an assistant to reverse the plug while you do the listening (Fig. 403).

Hum-balance control

Audio components, even those whose tube filaments operate on dc, frequently are equipped with a control whose purpose is to cancel hum originating in the tube filaments. You may find that

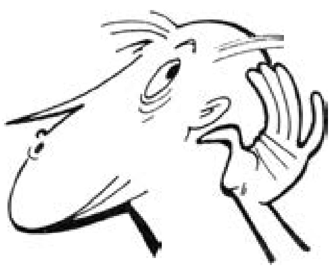


Fig. 403. *Listening for least hum.*

you can turn this control quite a way with little effect on hum level, but then you come to a small critical range where hum drops sharply and rises just as sharply. Accordingly, the adjustment of their control should be a painstaking one. You should repeat this adjustment after the component has been in use a while and after any tube has been replaced.

The best positions of the hum-balance control and of the line plug may be interdependent. After adjusting one, you should recheck the other.

Tubes

When hum originates in a tube, that tube is most apt to be in the first or second stage of the component in question. The instruction manual accompanying the component may contain a block diagram identifying the location of each tube stage. If not, your radio dealer or else the component manufacturer can supply this information.

Tube hum is most apt to be a problem when the filaments are operated on ac rather than dc. If you carry a stock of replacement tubes, you can check for a "hummy" tube in the troublesome component by replacing one tube at a time, preferably beginning with the first stage (where the signal enters). If you have several tubes of the type used in the first stage, try them all, because there can be quite pronounced differences among ac-operated tubes with respect to hum. Don't forget to adjust the hum-balance control when trying each tube.

A tube can pick up hum, not only from its own filament, but also from nearby sources of a 60-cycle magnetic field, such as transformers and motors. Therefore a new tube is occasionally the cure for hum even though the stage is operated on dc filament current. That is, one tube may be less susceptible than another to external hum sources.

While the tubes of an FM tuner are usually operated on ac, there isn't much danger of hum because the early tube stages, with one exception, reproduce radio rather than audio frequencies. The exception is the oscillator tube, where 60-cycle current leaking from the filament into other sections of this tube will be reproduced as hum. Hence if hum is proving bothersome in an FM tuner, you stand a good chance of curing the trouble by replacing

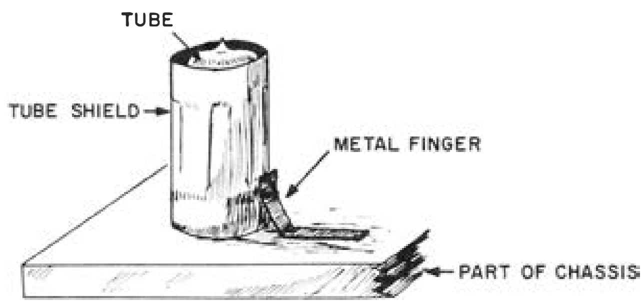


Fig. 404. *A means of grounding the tube shield.*

the oscillator tube. One way of gathering evidence against the oscillator tube is to note whether hum increases when you dial to *each* station and decreases when you dial "off-station."

When checking tubes by substitution, turn off the power before removing or inserting a tube. Otherwise there may be a surge of current that can harm the tube, the speaker or other parts.

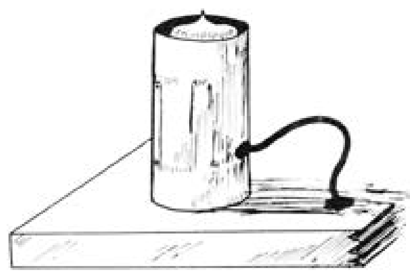
Tube shields

Motors, transformers, chokes, cables carrying alternating current, and other items produce magnetic fields that can induce hum in a nearby tube. Therefore the tubes in the early stages of a component—and sometimes all the tubes—are enclosed in metal shields to ward off such fields. A loose or missing tube shield can account for an appreciable amount of hum.

To do its job, the shield must not merely hang on the tube but must make firm contact with ground, that is, with the metal chassis of the component. Usually the tube socket is constructed so that it makes firm contact with both the chassis and the tube shield. Sometimes, as in Fig. 404, a metal spring finger extends from the chassis and contacts the shield. In either case, it is a simple matter for you to check whether the shield makes a secure connection with ground (chassis).

You may wish to put a shield over a tube that the manufacturer, for reasons of economy, decided to leave unshielded. If so, you will

Fig. 405. *Grounding a tube shield by soldering a wire between it and the chassis.*



have to figure out a way of connecting the shield to ground. Merely hanging the shield on the tube will probably increase the hum level rather than lessen it. A way of making this ground connection (Fig. 405), is to solder one end of a piece of wire to the tube shield and the other end to the chassis. The wire should be fairly heavy and as short as feasible (but not so short that you can't remove the shield without unsoldering the wire). If it isn't practical to solder the wire to the chassis, you can perhaps find a nearby chassis screw to which you can attach the wire. (Never shield a rectifier tube, because the shield will cause excessive temperature rise, due to trapping of heat, and the tube's life will be unduly shortened.)

Chassis covers

Components such as preamplifiers, power amplifiers, etc. typically have top and bottom covers, not merely for appearance and exclusion of dust, but also for shielding against hum fields. When a cover is removed in order to replace tubes or for any other purpose, be sure not only that the cover is replaced but also that the fastening screws are all put back and well tightened.

Ground to earth

Earth is part of the electrical path for the ac employed in homes and other structures. The chassis of a high-fidelity component is insulated from the house current (by a transformer or other means), but there is inevitably a small amount of leakage which permits a minute amount of 60-cycle voltage to appear between the chassis and earth. Depending how minute this voltage is, the chassis itself may be a source of hum.

This problem can be met, as shown in Fig. 406, by grounding the chassis to earth. This means connecting a relatively heavy

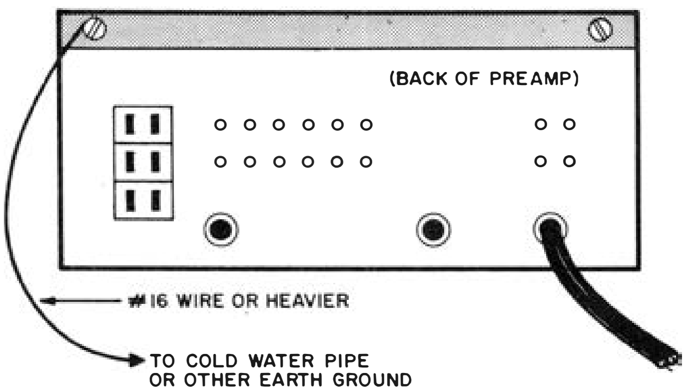


Fig. 406. *Grounding the amplifier.*

wire (No. 16 gauge or larger) from the chassis to a cold water pipe or some other metal object firmly embedded in the earth. Never make the connection to a gas line. To attach the wire to the chassis, loosen one of the screws that secures either the top or bottom cover. Make the earth connection to either the preamplifier or power amplifier, preferably the former.

Internal grounds

Audio components are usually connected to each other—for example, the tuner to the preamplifier—by a shielded cable, consisting of a solid or stranded lead within a tubular lead (Fig. 407). The inner lead is coated with insulating material, and the other lead usually has a protective covering of plastic, cloth or other substance. The inner wire is known as the "hot" or signal-carrying lead. The other one is called the shield or the ground lead, and diverts hum fields from the hot lead as well as making the necessary electrical connection between one chassis and the other.

An insecure ground connection between components can produce appreciable hum. A broken connection can produce a devastating amount of hum.

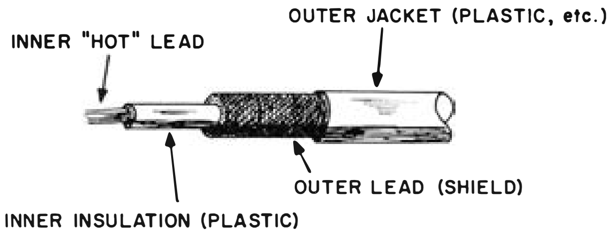
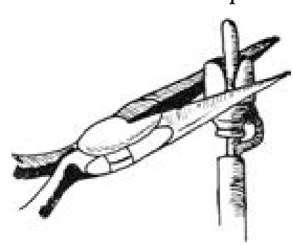


Fig. 407. *Shielded audio cable. See also Fig. 203 on page 17.*

The insecure ground connection may be due to a loose fit between the plug to which the shielded cable is attached and the jack into which the plug is inserted. A slight squeeze of the plug with a pliers (Fig. 408) may be enough to dispose of a hum problem. The poor connection may be due to dirt or rust on the plug or jack or both. Scraping or filing is needed here; or perhaps

Fig. 408. *Tightening a loose-fitting phono plug.*



replacement of the plug or jack or both. When connecting and disconnecting the plug from the jack, be sure that the audio equipment is shut off. Otherwise, you are likely to produce a tremendous pop or roar in the speaker, possibly damaging it.

The ground lead of the shielded cable may have a hairline, intermittent or complete break which remains invisible because of the outer covering. To check this possibility, try substituting another cable. Or test the original cable with your continuity checker (Chapter 2).

In the case of the turntable, its metal base and the frame of the motor present special grounding problems. If the tone arm is made of metal, then the arm also is part of the special grounding prob-

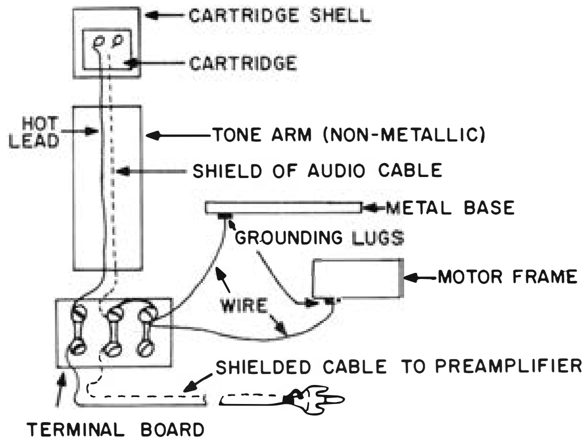


Fig. 409. *A method of grounding the metal base and motor frame of a phono turntable (unlikely to result in least hum).*

lems caused by what we will hereafter call "phono metalwork." Like any large mass of metal, the metalwork items pick up hum from the turntable motor, from nearby transformers and from other sources of an alternating magnetic field, such as wires carrying 117-volt house current. This hum must be grounded (short-circuited) to the preamplifier chassis to prevent the hum from affecting a sensitive magnetic phono cartridge.

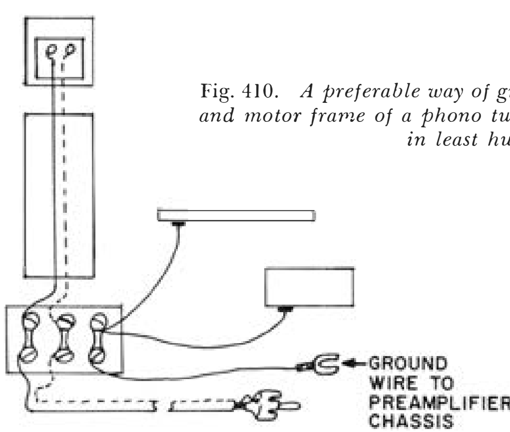


Fig. 410. *A preferable way of grounding the metal base and motor frame of a phono turntable (likely to result in least hum).*

It has frequently been the practice of phonograph manufacturers to ground the metalwork through the ground lead of the shielded cable issuing from the phonograph, as shown in Fig.

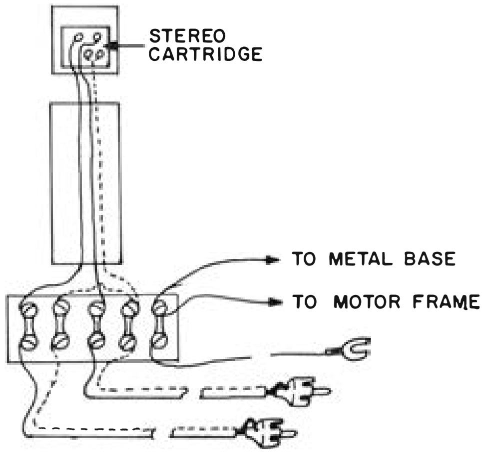


Fig. 411. *Three-wire phono system.*

409. Usually—the results of grounding techniques are not completely predictable—it is better practice to use a separate wire for grounding the metalwork to the preamplifier chassis, as in Fig. 410. Any pre-existing connection between the shielded cable and the metalwork must then be broken. When two ground paths exist between the phono cartridge and the preamplifier, they form what is known as a "ground loop," which is, in effect, a big one-turn coil of wire that picks up hum from surrounding magnetic fields.

Owing to the multiplicity of ground leads, hum problems tend to be more acute in stereo than in mono turntables. We have already mentioned one tactic for reducing hum — running a separate ground wire from the phono metalwork to the preamplifier chassis. Another is to convert from the three-wire system employed in a number of turntables (usually changers) to a four-wire system. In a three-wire system (Fig. 411), the two ground terminals of the stereo cartridge are connected to a single ground lead that runs through the cartridge shell and often through the tone arm as well. (Eventually, either in the tone arm or at the point of connection to the phono output cables, the single ground lead is divided into separate ground leads.) In a four-wire system, as shown in Fig. 412, separate ground wires are employed all the way from the cartridge terminals to the phono output cables.

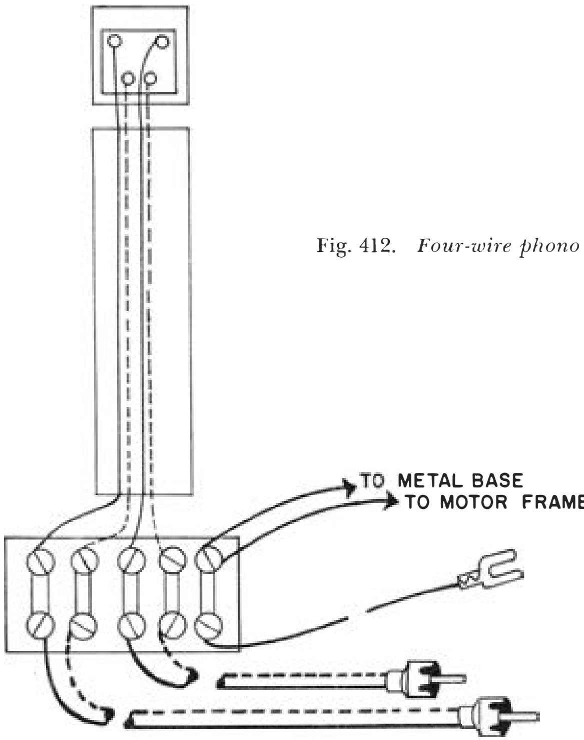


Fig. 412. *Four-wire phono system.*

A third hum-reducing measure is to twist the two shielded output cables around each other, as in Fig. 413.

While each of these strategies has a good chance of reducing hum, you can't be sure until you try. Hum problems can be quite tricky, and sometimes your efforts may even backfire and increase the hum level.

Similar problems are encountered in carrying the signal from a tape playback head directly to the preamplifier of your audio system. Again, try to discover whether hum is reduced by running a separate ground lead from the tape deck chassis to the chassis of your preamplifier, and by twisting the output cables about each other (in the case of a stereo tape head).

Cable routing

The shielded cable going from the phono cartridge or tape playback head to the preamplifier should be routed as far as possible from transformers and motors in the audio system. Although the quantity of hum picked up by the cable from a transformer or motor may be quite small, it is nevertheless apt

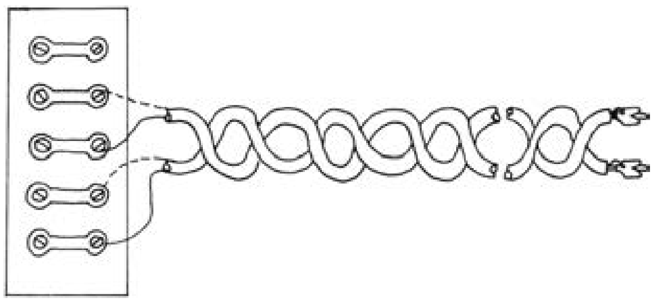


Fig. 413. *Twisting the output cables to cancel hum.*

to be sizable in relationship to the minute signal produced by the phono cartridge or, especially, the tape playback head.

Keep the phono or tape cable away from the ac line cord (Fig. 414). Sometimes there is a temptation to run the cable and cord close together for the sake of neatness. But here is one case where it does not pay to be neat.

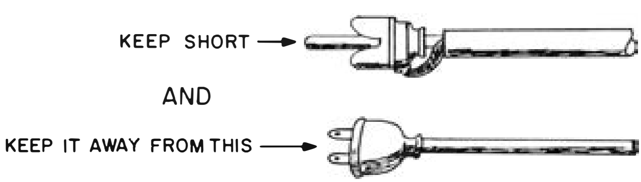


Fig. 414. *Two precautions against hum.*

A long run of cable, however carefully routed, may account for excessive hum pickup. So always use as short a cable as practical in connecting components. (As a bonus, this practice also helps to prevent or reduce treble loss due to cable capacitance.) If a connecting cable is appreciably longer than necessary, and you are disinclined to shorten it, *don't* form the excess cable in the shape of a coil, because this will increase its capability of picking up hum.

Location of components

You must place audio components judiciously with respect to each other to minimize hum. It is poor practice to place a phonograph close to a power amplifier, because the magnetic field emanating from the amplifier's power transformer may induce hum in a magnetic phono cartridge. If the preamp is located too close to or on the wrong side of the power amplifier, the early stages of the preamp may pick up hum from the power amplifier's transformer. In the main, it is the power amplifier that has to be kept fairly distant from, or at least judiciously oriented with respect to, the other equipment. Still, you should be alert to the possibility that any ac-operated component may induce hum in another component.

Hum induced by one component in another can sometimes be virtually eliminated by interposing a piece of silicon or mu-metal or sheet metal (silicon steel or soft iron is best) between the two. For example, I found in one problem situation that hum was being induced in the early stages of a preamplifier because they were located directly over the power transformer of an FM tuner. For all practical purposes, this hum was eliminated by placing a piece of sheet metal beneath the section of the preamplifier containing the early stages, as shown in Fig. 415.

Excessive bass

At the beginning of this chapter we told you that hum is always with us and the best we can hope to do is to keep it below audibility. However, improper operation of an audio system may exaggerate hum enough to make it audible. Specifically, too much bass boost may lift hum to the annoyance level in some systems. Perhaps you have turned up the bass control too much in relation to the volume at which the system is operating. The louder the volume, the less bass boost is ordinarily required to maintain apparent balance among the various frequencies. Perhaps your loudness control is misadjusted, supplying more bass boost than is required at low listening levels. Perhaps, in view of the efficiency of your speaker system in the bass range, you are better off without the loudness control than with it.

Wrong phono input

Various brands of magnetic phono cartridge differ considerably in their signal output. Therefore some preamplifiers have two input jacks (or two *sets* of jacks in the case of stereo units) for a magnetic cartridge: a "high" jack for high-output cartridges, and a "low" jack for low-output ones.

If you inadvertently connect a low-output cartridge to the "high" jack, the phono signal receives less amplification than it properly should. Perhaps you don't mind because you prefer to do your listening at moderate volume levels. However, you may be cheating yourself in terms of hum. The hum produced by the preamplifier remains the same, regardless which phono input jack is used. Therefore you will have a higher hum level relative to the audio signal than if you had connected the low-output cartridge to the "low" jack.

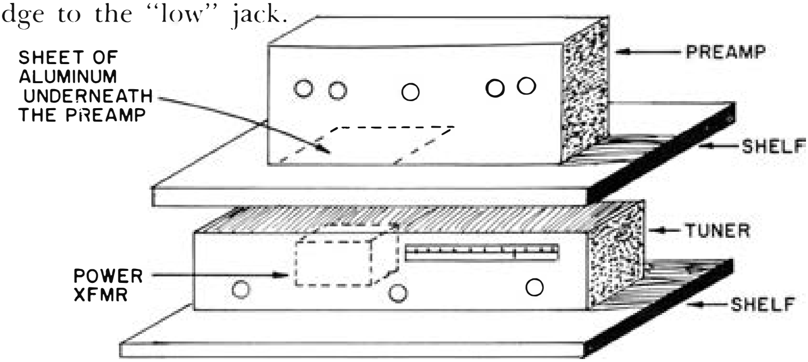


Fig. 415. *Eliminating hum pickup by using a piece of sheet metal as a shield between components.*

(If you are thinking of connecting a high-output cartridge to the "low" jack in order to maximize the ratio between audio signal and preamp hum, *don't!* This is apt to result in appreciable distortion because the first preamp stage is being overloaded by the high phono signal.)

Sensitivity of the pickup to hum

Magnetic phono cartridges differ not only in their signal output but also in their relative sensitivity to hum pickup. Hum sensitivity must be considered, not as an absolute quantity, but in relation to the amount of signal produced by the cartridge. Thus if cartridge A picks up twice as much hum as cartridge B, but A also delivers twice as much signal as B, then the two pickups have the same hum sensitivity in relative terms.

Hum sensitivity is one factor you should consider in choosing among pickups on the basis of listening tests. If possible, the test should be conducted with a turntable similar to or, preferably, identical with the one you will be using. Some phonographs provide better protection against hum than do others.

If you plan to do your own mounting of a tone arm on a transcription turntable board, be sure to follow the recommendations of the turntable manufacturer concerning location of the arm. Otherwise you may run into excessive hum owing to the relative positions of the pickup (in the tone arm) and the turntable motor.

Generally, the ceramic cartridges present less of a hum problem than do magnetic ones. They are by nature less sensitive to hum pickup, and they have an inherently much higher signal output.

Transformerless components

Occasionally an audiophile tries to incorporate in his audio system a component that has no power transformer to isolate it from the house current. For example, he'll try to connect an inexpensive AM radio or a transformerless TV set to the system. Usually this causes pronounced hum, and it may place a lethal voltage on the chassis of the other audio components.

Input level set

Power amplifiers frequently have an input level set to prevent excessive signal from being fed into the amplifier, with resultant

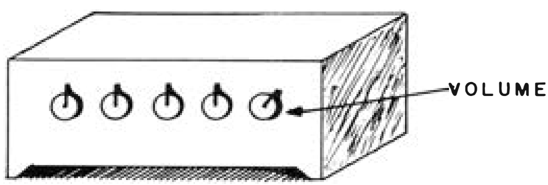


Fig. 416. *Volume control of the preamplifier in "2 o'clock" position.*

chance of distortion and perhaps damage to the speaker. As the level set is turned down, it reduces, not only the incoming audio signal, but also hum originating in the preamplifier. Accordingly, failure to turn down the level set may be responsible for excessive hum. If you get more than enough volume from your speaker long before the preamp volume control has been turned to about the 2 o'clock position (Fig. 416), this indicates that you should turn down the power amplifier's input level set. (However, don't turn it too far down, because this would require the preamp to produce correspondingly more signal, with increased chance of noticeable distortion.)

Preamps, too, frequently have input level sets to control the incoming signal from the tuner, phono pickup, etc. In this case, excessive hum may result from turning the level set too far *down* rather than up. The lower the setting of a preamp level set, the higher must be the setting of the preamp's volume control to achieve the desired sound level. Accordingly, there will be greater amplification of hum produced in tube stages between the level set and the volume control.

Program hum

Sometimes you will encounter substantial hum on an FM program, but the hum will die when another program comes on or when you turn to another station. The fault then lies with the station, and there is nothing you can do except complain to its management.

It is an unfortunate fact that a number of FM stations are prone to transmitting hum. Some have only occasional lapses. Others, because of carelessness, indifference or poor equipment, broadcast many hours of hum-ridden programs, particularly noticeable when you turn the volume high. Check this possibility out too before you condemn your FM tuner and the manufacturer who made it.

Similarly, phono discs and prerecorded tapes may contain hum.

Chapter 5
Noise Problems

THE PLEASURE YOU OBTAIN FROM REPRODUCED MUSIC IS GREATLY enhanced when the music issues from an utterly quiet background, unmarred by hiss, rushing, buzz, clicks, whistles, whooshes, snaps, pops, squeals, crackling, howling, sputtering, frying, crosstalk and other unwanted sounds that come under the heading of noise.[1] Much of the engineering that goes into a high-fidelity component has to do with the suppression of noise.

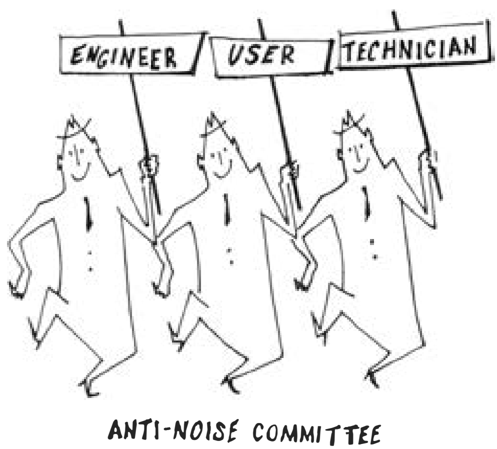


Fig. 501. *An important alliance.*

But it isn't only up to the audio engineer (Fig. 501) . Preventive and remedial steps must be taken by the user and by the audio technician. Many of these steps are simple enough to be taken by the user with little or no technical knowledge.

[1]Noise of course also includes hum, but this is an entire subject in itself and therefore is separately treated in Chapter 4.

Tube noise

Tubes are always and rightly a prime suspect when noise develops. Suspicion usually focuses on the first stage of a preamplifier, power amplifier, tape recorder or tuner—the stage where the audio signal enters (Fig. 502). Along with the signal, noise in the first stage is passed along to and amplified by all the following stages—wherefore the first-stage tube must be the best of the lot. If you are the sort of audiophile who thinks ahead, you will have in reserve several tubes of the type used in the first stage. If the need arises, you then have a choice of several tubes in striving for minimum noise. This follows the practice of various manufacturers who employ "selected" tubes to obtain maximum performance from their product. (The leftover tubes of the type employed in the first stage can usually be used as replacements for later-stage tubes in the same or in another component.)

Of course, a tube beyond the first stage may be responsible for noise. Assuming you have a full stock of replacement tubes, the

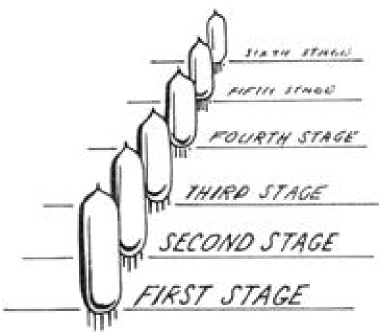


Fig. 502. *The order of suspicion where tubes are concerned.*

logical course is to replace tubes one at a time, working from the first stage to the last if you can identify their sequences. For some types of noise you can identify the offending tube by tapping each one lightly with the rubber end of a pencil; the offender will produce noise with each tap (Fig. 503).

Once in a great while, the culprit turns up elsewhere than in the component which appears to be giving trouble. The following case history illustrates the point: A preamplifier was intermittently squealing, sputtering and spitting. When the preamp was taken out of the audio system and the tuner was connected directly to the power amplifier, the symptoms vanished. Seemingly, the preamp was at fault. But replacement of every tube in the preamp did not help. On a hunch, tubes in the power amplifier were checked by substitution. When one of the output tubes was replaced, the preamp noise disappeared (the preamp had been put back in the audio system). What had happened? The preamp obtained its high-voltage dc requirements from the power amplifier. The defective output tube had an intermittent near-short, not profound enough to produce noticeable noise directly in the power amplifier. But

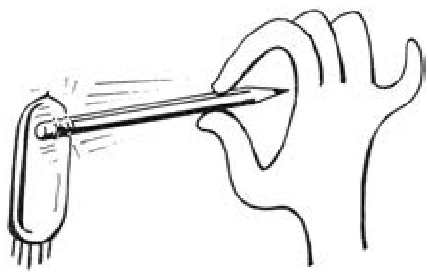


Fig. 503. *Tapping a tube with the rubber end of a pencil.*

the near-short did cause sudden, slight brief changes in the power drawn by the output tube, and these changes were transmitted via the common power supply to the preamp. To draw a parallel, think how your AM radio produces a click when you switch a light. The light and the radio are on the same power line, and the sudden current surge when the light goes on or off is reflected in the radio.

Noisy controls and switches

Volume controls can become nasty noise makers. As the control wears with use, electrical contact between its elements grows imperfect, and noise is the consequence as you turn the control. In fact, electrical contact may become so tenuous that intermittent noise occurs even when the control is let alone. The same may happen with the bass, treble and other continuous controls, particularly if frequently used. Switches too. Special cleaning fluids that not only dissolve dirt and film but also promote electrical contact are available at audio stores and electronic supply houses. Sprayed or inserted by eye dropper into the control or switch, these will usually restore it to satisfactory operation for a number of weeks or months. It will probably be necessary to remove a top or bottom cover from the component in question to get at the noisy part.

Eventually it will be necessary to replace this part, and you must decide whether you or an audio technician is going to do it. Generally it is fairly simple to replace a volume or tone control, especially if you have had kit-building experience and therefore can unsolder and resolder three or four wires connected to the control. But a switch usually is a more complex task because of its intricate wiring. Also, the switch is apt to be a special design obtainable only from the component manufacturer. If you attempt to replace a control or switch, before anything else be sure to draw a dia-

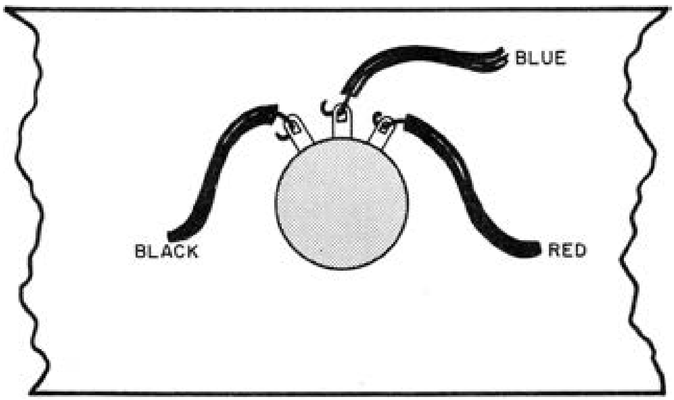


Fig. 504. *Audiophile's drawing of where the volume control leads go, prior to replacing the control.*

gram (Fig. 504) of all leads, resistors, capacitors and other parts connected to the item being replaced.

Noisy resistors

High-quality electronic components employ special low-noise resistors in the first stage and sometimes in the second stage as well. Unfortunately, a low-noise resistor sometimes changes its characteristics under the strain of use. Also, and unfortunately, these resistors cost roughly 10 times as much as the garden variety. The only way to test for a bad resistor is by substituting a new resistor, which can be rather an expensive proposition inasmuch as truly quiet resistors cost between $1.50 and $3.00 each at retail. Accordingly, if you have reason to suspect that a resistor has gone bad (tubes having already been checked out), your best bet may be to turn the problem over to an audio technician even though you have enough technical knowledge to replace resistors yourself.

On the other hand, satisfactory quieting is sometimes obtainable from a garden-variety resistor provided that it has excess wattage rating. For example, a 2-watt resistor may prove much quieter than the conventional ½-watter; and it is much less expensive than the low-noise types. If you are determined to have a go at replacing resistors yourself, and if you know your way around circuits a bit, devote your attention mainly to the load resistors connected to the plate and cathode of the tube in question.

Poor connections

In searching for the cause of noise, keep your eye out for insecure connections. Loose-fitting shielded cables are one item. When leads are attached to screw terminals, as in the case of the FM antenna and the speaker, check that the screws are tight and that stray strands do not form a minor or intermittent short between terminals (Fig. 505). A crackling or buzzing sound is sometimes traceable to an insecure connection between the phono cartridge terminals and the lugs that slip onto these terminals. To insure a tight fit, remove the lugs and pinch them slightly with a fine pliers. You might also try carefully cleaning the cartridge terminals. Similarly, there must be a good connection between the tone arm and the plug-in shell that holds the cartridge. Clean the contacts coming out of the shell, and be sure to insert the shell as far as it is meant to go into the tone arm.

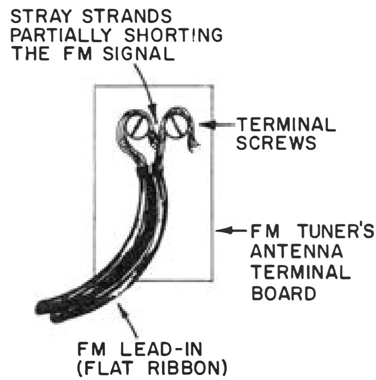


Fig. 505. *Poor connection of the FM antenna lead-in to the FM tuner, resulting in increased noise. See also Fig. 911 on page 127.*

Mismating of components

Mismating of components can account for noise attaining a higher level than it rightfully should. Take the case of a power amplifier and speaker that are mismated in terms of the amplifier's rating and the speaker's efficiency. The higher the wattage rating of an amplifier, the more noise it tends to produce, *everything else being equal.* (Of course, through superior design, it is possible for a high-wattage amplifier to produce even less noise than a low-wattage one, but *on the average* the high-wattage ones generate more noise than the low-wattage ones.) The more efficient the speaker, the more loudly will it reproduce the amplifier's noise. As a rule, high-power amplifiers are intended for relatively inefficient speakers, while low- or medium-power amplifiers are meant for relatively efficient speakers. If you use a high-power amlifier with an efficient speaker, you risk unnecessary noise—unnecessary because you may be using only a small fraction of the power which the amplifier can deliver, yet you will reproduce its noise to the full.

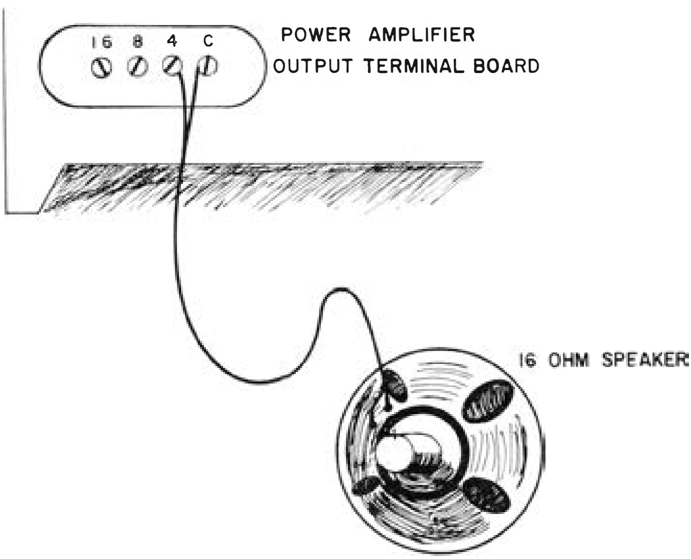


Fig. 506. *Connecting a 16-ohm speaker to the 4-ohm tap of a power amplifier to reduce the amount of power drawn and hence the reproduced amplifier noise.*

If the power amplifier is capable of much more volume than you need, and amplifier noise is objectionable, a partial or complete solution is to connect the speaker to an impedance tap on the amplifier lower in value than the rated impedance of the speaker. This will reduce the amount of power drawn by the speaker. For example, as in Fig. 506, you might connect a 16-ohm speaker to the 4-ohm tap of the amplifier. An alternate but less desirable solution is to put a noninductive resistor in series between one of the amplifier's output taps and the speaker, as in Fig. 507. The resistor should be between 1 and 3 times the speaker impedance and should be rated at 10 watts or more. Thus you might use a 30-ohm resistor with a highly efficient 16-ohm speaker. However, such a resistor interferes with the ability of the amplifier to "damp" the speaker. Damping makes for clean, tight bass reproduction instead of blurry or boomy bass.

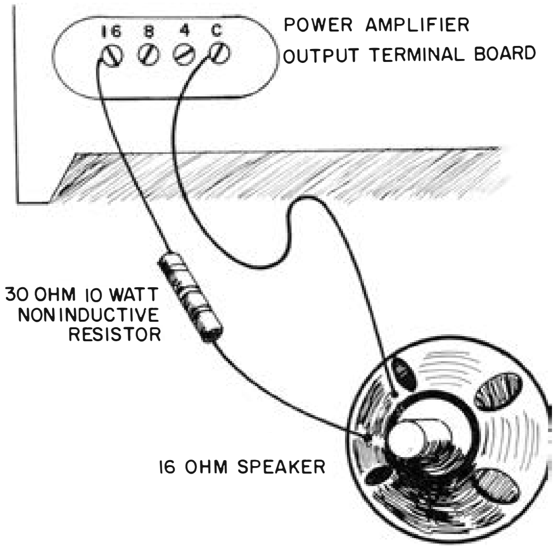


Fig. 507. *Using a series resistor to reduce the power drawn by a speaker.*

In similar fashion, the preamplifier may deliver too much audio signal and also too much noise signal to the power amplifier. The greater the gain of the preamp, the more noise it is apt to produce. Many power amplifiers contain an input level set that enables you to reduce the signal and noise received from the preamp. This should be used judiciously. If you turn it very far down to reduce noise generated by the preamp, you may make the preamp work extra hard to produce enough audio signal to drive the power amplifier—the result: appreciable distortion. As a rule, the preamp is not working too hard if you obtain moderately loud volume at about 2 o'clock setting of the preamp's volume control. If you get substantial volume much before 2 o'clock and if the power amplifier does not have an input level set to reduce the signal, a technician can install one for you.

Tuner problems

A variety of noise problems are indigenous to the FM tuner. Perhaps the prime one is that of a good antenna in fringe and sub-fringe areas, especially for noise-free reception of multiplex (stereo) programs, where the signal is inherently much weaker than on mono. For stations less than 15 miles away, a good tuner generally permits you to get away with almost anything in the way of an antenna. Up to about 25 miles, you can usually make a broad-band TV antenna serve satisfactorily. But when you reach much farther out, you risk substantial noise unless your antenna is a high-gain unit (Fig. 508) specifically designed for FM.

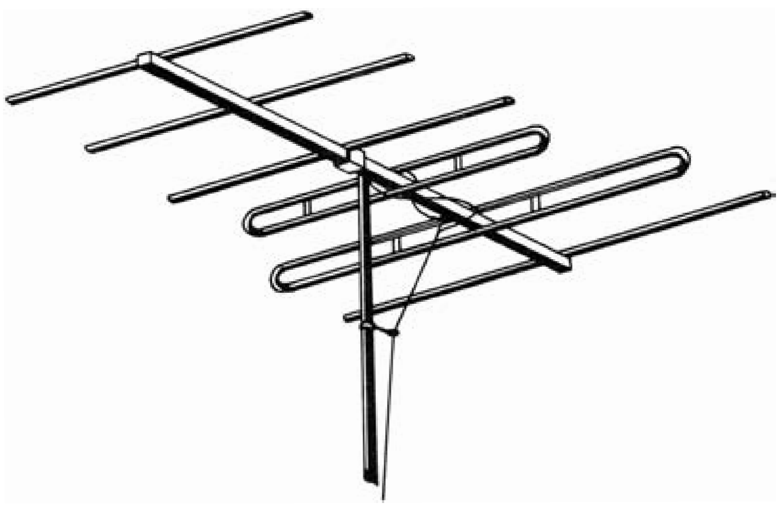


Fig. 508. *A high-gain FM antenna.*

If you install the antenna yourself, make certain that you know which end is supposed to face the station. Also be certain of the exact direction of the stations you want to receive. An antenna rotor may be needed to bring in several distant stations at well separated points of the compass. If you use a very high-gain antenna, a rotor becomes all the more necessary, because antennas become more directional as their gain is increased. It is a general rule that reception improves as you increase the height of the antenna. However, this rule too has its exceptions, and occasionally it is found that, within a range of several feet, a lower location gives better results than a higher one.

The elements conspire against good reception. Wind may loosen the antenna rods, resulting in signal loss and consequent noise in the tuner. Signal loss results from ice formation, rain, rust on the antenna, and rust at the connection to the lead-in wire. Obviously, some of these things cannot be helped, while others can.

The lead-in wire, which connects the antenna to the tuner, is nearly as important as the antenna. It should be as short as circumstances permit, because every foot cuts down the signal and increases the possibility of perceptible noise. Excess length should be cut off; in any case, it should not be formed into a coil, which inhibits passage of the signal. To minimize signal loss, the lead-in should be kept away from metal bodies, such as rain gutters. And sharp turns should be avoided.

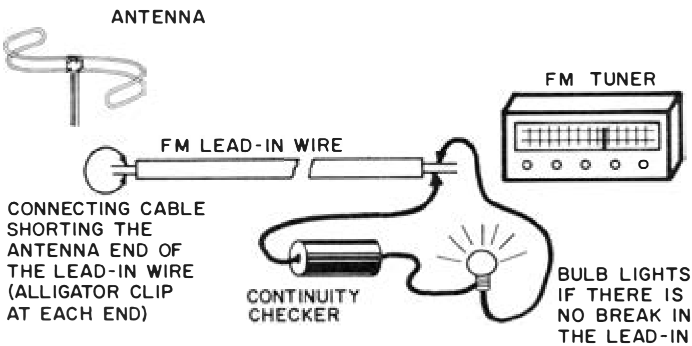


Fig. 509. *Checking the FM lead-in for a break.*

Last and certainly not least, the lead-in should be suitable for FM use. Usually, TV flat ribbon is correct. If shielded cable is employed, to minimize pickup of ignition noise and other interference, make sure that a type appropriate for FM is used. Find out whether you need a matching transformer between this cable and your FM tuner to obtain maximum transfer of signal to the tuner.

A slight break in the lead-in wire, quite possibly hidden from view, will impair reception. You can discover whether a break exists by disconnecting the lead-in from the tuner, shorting out the end that is connected to the antenna (someone may have to get on the roof to do this), and applying your continuity checker to the free ends of the lead-in, as shown in Fig. 509.

Passing vehicles may produce ignition noise. The answer is a good antenna mounted as high and far from the traffic as feasible.

But the lead-in may be acting as an antenna and picking up the interference. Then the lead-in should be shielded cable of an appropriate type; however, unless you use an expensive grade, shielded cable produces more signal loss than flat ribbon.

Refrigerators, oil burners and other electrical devices may produce clicks or pops as they go on and off, or frying and buzzing sounds as their motors run. This requires an electrical filter across the power line, mounted at the source of the difficulty by a qualified person. The filter can be purchased at an electrical or electronic supply house. In its simplest form, a filter consists simply of a 600-volt capacitor, with a value between 0.01 and 0.1 μf, mounted across the line, as shown in Fig. 510. More complex filters may be required in stubborn cases of noise produced by electrical devices.

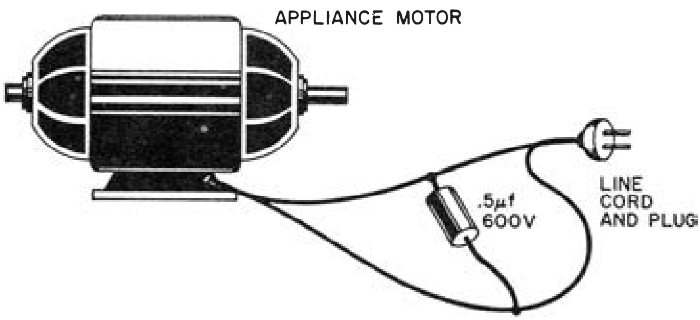


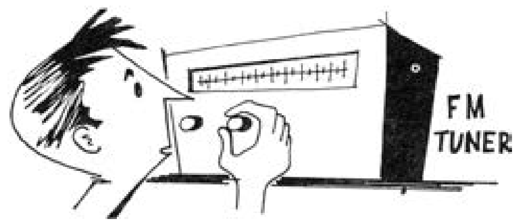
Fig. 510. *A simple filter to eliminate noise caused by an electrical appliance.*

A little sleuthing is sometimes necessary to locate an offending electrical device. Thus the culprit may be a defective light switch, a light bulb barely hanging on to life, or a light bulb insecurely screwed into its base. In one instance, intermittent buzzing in the FM tuner was traced to a faulty pilot lamp in the tuner itself.

Accurate alignment of the FM tuner is vital for noise-free reception of distant stations (and for minimum distortion). This is strictly a matter for the audio technician, and only a highly qualified one at that. The technician who lacks the proper equipment and experience, or who attempts to align by ear, can do your tuner more harm than good. So be careful as to whom you turn over your tuner to have its alignment checked and adjusted. You may find it wise to write to the manufacturer of your tuner for a recommended service agency in your locality. On the other hand, some inexpensive tuners are not susceptible to good alignment, no matter how much effort and skill are used.

If the antenna, lead-in and tuner are all ship-shape, the only deterrent to noise-free reception (distance permitting) is your failure to tune accurately to the desired station. Even though your tuner has afc (automatic frequency control) or a wide-band detector or both, it is desirable that you tune as precisely as possible to the desired station, particularly if it is a distant one (Fig. 511).

Fig. 511. *Tune carefully!*



Most tuners drift somewhat during the first 10 minutes or more of operation, so readjust the tuning after the warmup period.

Phono problems

The principal indigenous cause of noise in a phonograph is apt to be the grit, dust and electrical charges in the record groove. Various special items to remove these are available at audio supply

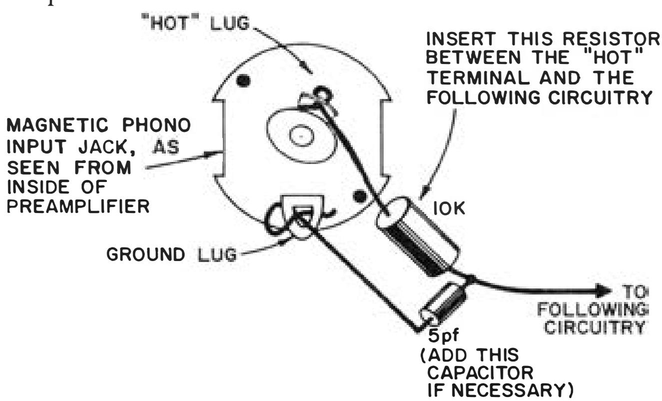


Fig. 512. *Preventing pickup of radio stations by the phono amplifier.*

houses and record stores. Most are liquids for application to the disc before playing; be sure to allow sufficient time for the disc to dry if you apply one of them. A few are devices affixed to the tone arm or turntable base that clean the disc as it is played.

A close-by radio station is occasionally picked up by a phono amplifier; such interference comes under the heading of noise. The cure is usually simple: Connect a resistor of about 10,000 ohms between the phono input jack's hot terminal and the following circuitry (Fig. 512) . Again, this is a situation where the individual with kit-building experience can probably handle the problem, or else a service technician can probably dispose of your problem in a jiffy. If the 10,000-ohm resistor does not produce adequate results, try connecting a 5-pf capacitor between the far end of this resistor and the ground terminal of the phono input jack, as in Fig. 512.

When the phono system is operated at a loud level, acoustic feedback can cause a howling or roaring sound, as illustrated in Fig. 513. Sound waves produced by the speaker may cause mechan-

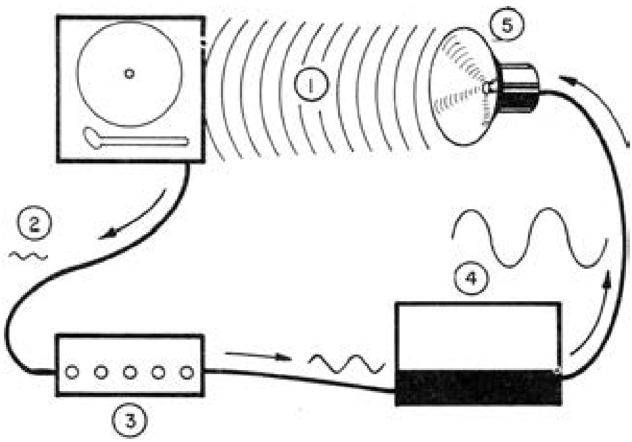


Fig. 513. *Acoustic feedback.*

ical vibration somewhere in the audio system—in a tube element or in part of the phono mechanism, such as the tone arm or cartridge. The vibration is electronically amplified by the audio system and issues from the speaker in the form of sound waves, which cause increased vibration, and so forth, culminating in an unpleasant sound. The cure may be to put greater distance between the speaker and the rest of the audio system—perhaps the speaker should be removed from a cabinet or shelf space shared with other components. If the phonograph is in a separate cabinet, closing the cabinet door part way or completely during operation may effect a cure. Mounting the phonograph base on a thick pad of foam rubber may help. Since acoustic feedback may be due to an unusually vibration-sensitive tube in the phono stage of the preamplifier, try a substitute tube here.

For proper treble response, a magnetic cartridge must be loaded with a resistor of suitable value, specified by the cartridge manufacturer. This value differs somewhat among cartridge brands and sometimes among different models of the same brand. Too high a value will result in a treble peak, accentuating record noise. Therefore you should check with your audio dealer, and possibly with the manufacturers of your cartridge and preamplifier, whether your preamplifier provides the correct load resistance for your cartridge. If the value is wrong, a technician can alter the load resistance in short order. Or, if you have the skill of a kit builder, you can make this alteration yourself by replacing one resistor. This load resistor is usually mounted between the hot and ground terminals of the magnetic phono input jack, as shown in Fig. 514.

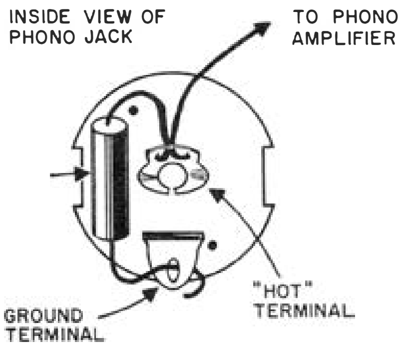


Fig. 514. *Customary location of the load resistor for magnetic cartridges.*

Noise is occasionally due to a physical malfunction that prevents the turntable from spinning truly. For example, an audiophile complained that many of his records sounded as though they were scratched. Once a revolution there was the familiar click associated with a scratched disc. Also, the stylus sometimes skipped grooves. It was found that a small lump of gummy dirt had formed on the inside rim of the turntable, which was driven by a rubber idler wheel. When the wheel hit the dirt spot, which was once every revolution of the turntable, speed changed briefly and sharply.

Tape recorder problems

The list of noise problems native to a particular component is especially long and varied in the case of the tape recorder.

Tape heads gradually become magnetized with use, producing noise in playback and also causing noise to be recorded on the tape. Therefore it is important to demagnetize the heads periodically, say after about 8 hours of use. Head demagnetizers, such as the one in Fig. 515, are inexpensive enough. Tape guides and other metal parts of the machine contacted by the tape should also be demagnetized. Of course, the tape machine's power should be shut off during this process.

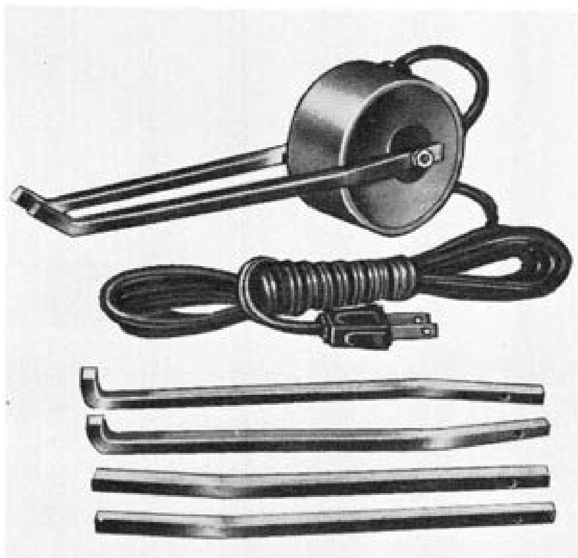


Fig. 515. *Tape-head demagnetizer.*

The tape oscillator is one of the principal causes of noise in recording, owing to distortion in the waveform of the bias current supplied to the record head. You can easily check whether waveform distortion is causing substantial noise. Put several feet of virgin or bulk-erased tape through the recording process but with no signal input and with the recording gain control all the way down. Play back the tape. If the "recorded" portion of the tape is much noisier than the unrecorded portion, the fault is the oscillator waveform. Try replacing the oscillator tube. If this doesn't help, and you find the oscillator noise objectionable, the problem belongs in the hands of a qualified technician. (Of course, it is possible that the design of the machine doesn't permit any improvement.)

Incomplete erasure of a previous recording is a form of noise. The fault may be yours because the previous recording was made at too high a level, so that you have no right to expect the erase head to cope with the abnormal task presented to it. Then you need a bulk eraser.

The erase head may be responsible in one of several ways: (1) The head may be defective. (2) Oscillator current supplied to the head may be insufficient. (3) The oscillator frequency may be too high for the head to operate efficiently. (4) The erase head may be misaligned with respect to the record head, so that the gaps of the two heads don't span the same portion of the tape (Fig. 516).

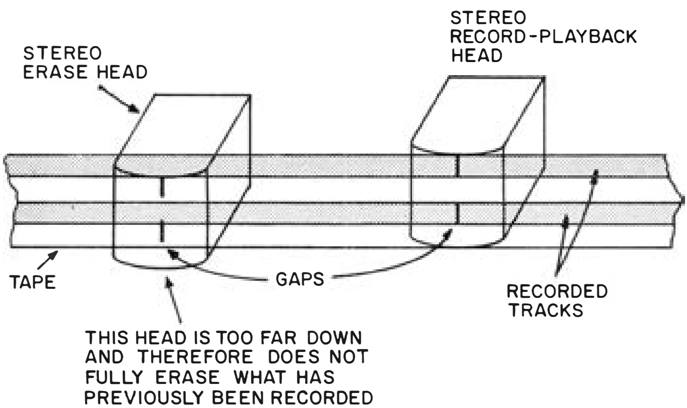


Fig. 516. *Mispositioned erase head.*

If the sound remaining after "erasure" contains the full audio range, the trouble is probably due to inexact correspondence between gaps. By following the instructions in the service manual, you may be able to position the erase head satisfactorily. But if the remnant noise consists mainly of low frequencies, which are the hardest to erase due to their deep penetration of the tape, the cause is probably one of the others listed above and you need a technician.

If a stereo recording head or playback head (or a single head used for both recording and playback) is positioned too high or too low, this may result in crosstalk because the signal intended for one tape track may overlap onto the adjacent track. The service manual will tell you how to reposition the head, but this job is usually best left to the audio technician. After a head is moved, it is important to adjust its azimuth alignment; that is, make sure the gap is exactly perpendicular to the long dimension of the tape (Fig. 517) . Otherwise treble response will suffer. Azimuth alignment requires a test tape. Adjust the playback head for maximum response while playing this tape. If the tape recorder employs a separate record head, then adjust the record head for best treble response when recording a fresh tape.

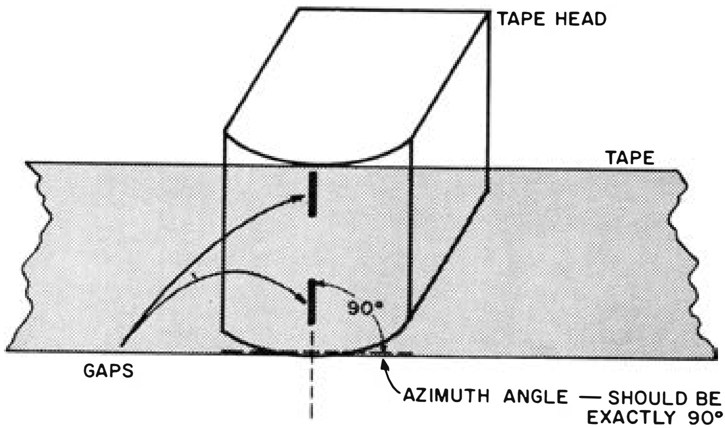


Fig. 517. *Azimuth of a tape head.*

Print-through—the transfer of loud sounds on one layer of tape to adjacent layers—may result from recording at an excessive level. At normal recording level it may be due to the use of tape with an excessively thin backing, such as ½-mil tape.

As the tape moves past the heads, it may squeal due to the dry condition of the tape or rough condition of the pressure pads (usually employed in home machines) . Squeal can be recorded on the tape, spoiling a valued recording. Quality tapes made by reputable firms are one answer to this problem; they incorporate lubricants to smooth the passage of the tape over the heads. You can apply a lubricant, specially made for the purpose and available at audio stores, to the heads, pressure pads and guides, but *not* to the capstan and pressure roller. Cleaning the heads and guides with alcohol will also help; clean the capstan and pressure roller too. At least one preparation on the market applies lubricant to the tape itself as it moves between reels. Replace the pressure pads if they appear to have lost their softness and smoothness. Moisture can be restored to a dry tape by storing it for about a day in a closed box containing a moist sponge.

If you record much below the level which is permissible from the distortion viewpoint, tape hiss plus the noise produced by the playback amplifier will be unduly high compared with the recorded signal. No home tape recorder, or professional machine for that matter, has decibels to spare where noise is concerned. On occasion, a faulty record-level indicator will tell you to record at a lower level than is really necessary. Check this out by making several recordings at successively higher levels and noting which of these first shows noticeable distortion. Check the corresponding position of the record-level indicator. If you have reason to believe that the record-level indicator is miscalibrated, turn your problem over to a technician.

A boggled splice is apt to produce an unpleasant thump, click, pop or wavering sound as the spliced portion goes past the playback head. A proper splice is one where the magnetic tape is cut at a 45° angle and the splicing tape is applied at the same angle (Fig. 518). This permits the magnetic and splicing tapes to make gradual

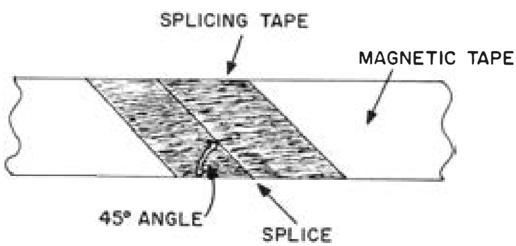


Fig. 518. *A correctly spliced tape.*

contact with the playback head and with the pressure pads (if any), avoiding splicing noise. As you probably know, the splicing tape is applied to the back (shiny side) of the magnetic tape.

Multiplex interference is a relative newcomer to the list of tape noises. A multiplex tuner or adapter produces a frequency of 19,000 cycles plus its harmonics (multiples of 19,000 cycles). One of these harmonics, if sufficiently strong, may "beat" against the tape oscillator frequency, generating an audible spurious signal. For example, an appreciable amount of 76,000-cycle harmonic might get into a tape recorder with a bias frequency of 70,000 cycles. The difference between these two is a beat frequency of 6,000 cycles, which will be audibly recorded on the tape. The solution is to have a filter installed between the tuner's output and tape recorder's input .Such filters (e.g. the Viking MX-10) are now commercially available, and you can install one yourself in a few moments.

With certain tape recorders and preamplifiers, electrical feedback can occur. Most home tape machines use the same tape head for both recording and playback, and therefore they have only one amplifier for both purposes. In some of these units, the output jack remains connected to the tape amplifier in the record mode.

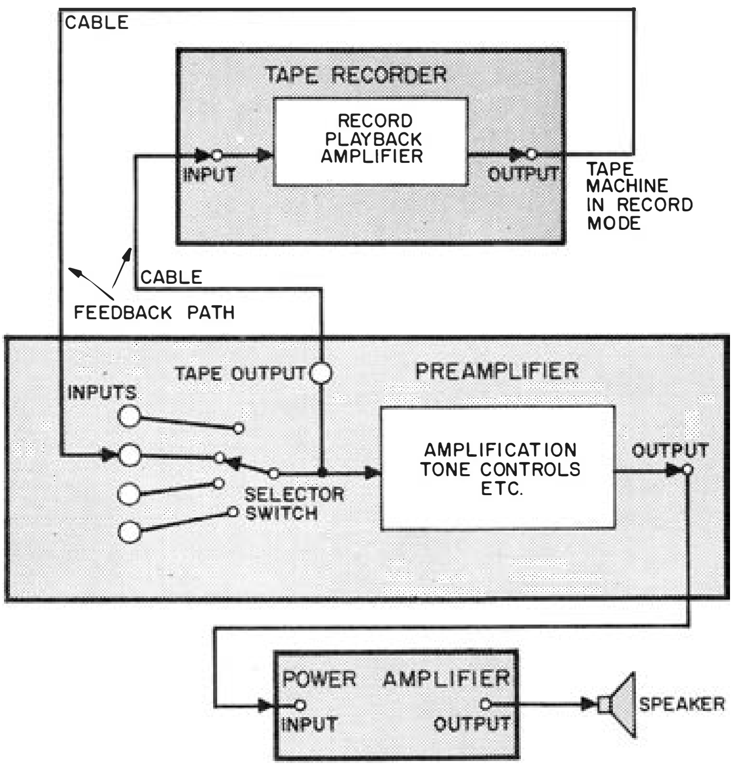


Fig. 519. *How feedback can occur when using a tape recorder.*

If the tape machine is in the record mode and the selector switch of the preamp is accidentally set to tape playback, the sequence of events (Fig. 519) may be this: Any random signal entering the machine's input jack is amplified, goes to the machine's output jack, proceeds via cable to the tape input jack of the preamp, goes to the preamp's selector switch and then its tape output jack, re-enters the input jack of the tape machine, is further amplified, and so on, culminating in a violent howl or screech.

The obvious answer is not to turn the preamp selector switch to tape playback when the tape machine is in the record mode. But accidents do happen, particularly if you shift frequently between the record and playback modes, as you might do at a party where you are recording tapes via microphone and playing them right back for the amusement of your guests. A better solution is to make sure when purchasing a preamp that it makes provision for breaking the feedback path. Such a preamp has a special input jack for tape playback, and a special "tape-monitor" switch that bypasses

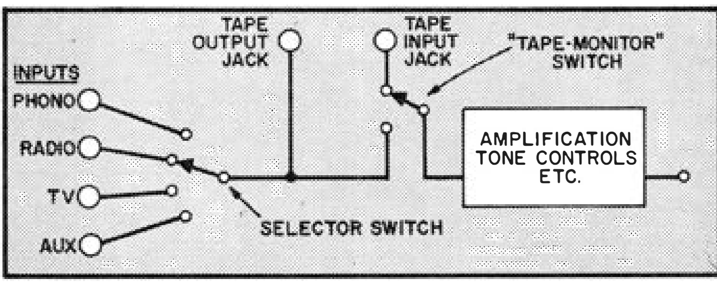


Fig. 520. *Preamplifier with a "tape monitor" switch which prevents electrical feedback.*

the regular selector switch (Fig. 520) . Alternatively, when buying a tape recorder you can choose one that includes provision for interrupting the feedback path.

Excessive treble

Anything in the audio system that exaggerates treble response will, by the same token, exaggerate noise. Thus a cartridge or speaker with a pronounced treble peak will produce more apparent noise than a smoother cartridge or speaker. Failure to set the preamp equalization controls correctly when the signal source is a magnetic cartridge or tape head, or exaggerated setting of the treble control, will emphasize noise. Use of a presence control may do the same, because the control accentuates frequencies in the neighborhood of 3,000 cycles, where noise is highly apparent to the ear.

Chapter 6
Bass Problems

IN THE STRUGGLE TOWARD HIGH-FIDELITY REPRODUCTION, A MAJOR objective has been to encompass the extremes—namely the very low and the very high frequencies—of the audio range. Full bass imparts body and warmth to the sound; full treble imparts naturalness and clarity. Each of the frequency extremes has its special problems—therefore the present chapter deals only with the question of bass, and treble response is treated separately in Chapter 7.

While extended bass reproduction is inherently desirable, you can have too much of a good thing. Excessive bass tends to impart a muddy or boomy character to sound. Also, it exaggerates undesirable sounds such as hum and rumble, which every audio system has to contend with. (Even if you don't own a phonograph, rumble is sometimes transmitted by an FM station when it is playing phono discs.) Therefore in discussing bass problems we are perhaps just as concerned with those involving excessive bass as with those involving too little bass.

Speaker problems

Speakers are mostly sold today as integrated systems, that is, the speakers are already mounted in an enclosure specifically designed for them. Typically, the integrated speaker system includes a woofer and tweeter that divide the audio range between them. For example, the woofer may reproduce the frequencies below 800 cycles, while the tweeter reproduces those upward of 800 cycles. Less often, the audio range is divided into three parts instead of two, and a mid-range speaker is included in addition to a woofer and tweeter. Thus the woofer might cover up to 800 cycles, the mid-range unit (sometimes called a squawker) 800 to 5,000 cycles, and the tweeter upward of 5,000 cycles. For convenience, the following discussion is limited to a woofer–tweeter combination, but it should be understood that it also pertains to systems that include a mid-range unit.

Manufacturers of integrated speaker systems frequently include a balance control for varying the tweeter level with respect to the woofer. Usually they mark the control setting that normally achieves acoustic balance between the tweeter and woofer. However, the user may accidentally have set the control too high or too low, resulting in the bass portion of the audio spectrum being relatively too weak or too strong. Some audiophiles even set the control to maximum deliberately, apparently in the misguided belief that this makes for highest fidelity.

The best way to set the speaker balance control is by ear—your ear. Judge for yourself what sounds like normal musical balance, the kind you hear in the concert hall rather than what you may have heard in an audio salon or in a friend's home. If you don't trust your ear, you can set the control to the "normal" mark indi-

Fig. 601. *Example of a room furnished in a "hard" manner.*

cated by the manufacturer. But bear in mind that if yours is a "hard" room (Fig. 601) containing much less than the usual amount of carpeting, drapes and soft furniture, this may accentuate the treble range, and the tweeter control really should be set somewhat below the normal mark. Conversely, if yours is a "soft" room, as illustrated in Fig. 602, with more than the usual amount of carpeting, etc., so that the high frequencies are virtually soaked up, the balance control should probably be set above the normal mark.

(Some dedicated audiophiles go so far as to adapt the furnishings of the room to the speaker and to their taste in acoustic balance. This is mentioned only to indicate the importance that room furnishings play in the quality of sound obtained from the speaker.)

Fundamentally, bass reproduction depends more on the speaker than on any other component in your audio system. Inasmuch as

Fig. 602. *Example of a room furnished in a "soft" manner.*

a given speaker may sound different in different environments, it is a good idea when choosing a speaker to do your listening in more than one audio store. When you have found a speaker that you *think* you like best (for the money), find out how it sounds in another store. You may discover that your preliminary choice is not all you thought it was.

Some stores permit you to exchange your purchase for another speaker if it turns out that your original choice doesn't sound as good in your home as it did in the store. In fact, some stores will even sell you a speaker on a money-back basis, except for a small charge to defray their costs. These privileges are worth seeking out.

You may find that the speaker's ability to reproduce bass is better in one part of the room than in another. A corner location, in particular, is apt to result in fullest bass, because the sound is concentrated into a 90° angle toward the listener. If the speaker is against

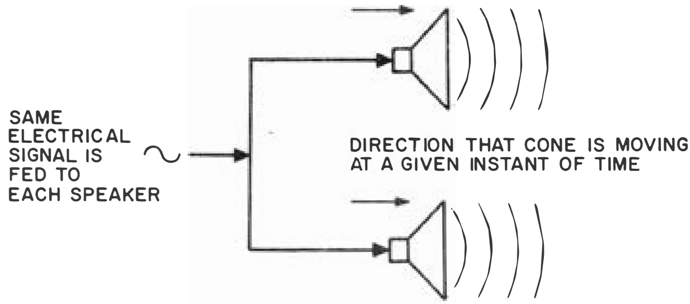


Fig. 603. *Speakers operating in phase.*

a wall, the sound is dissipated over a 180° angle, much of it away from the listener. In this connection, remember that low frequencies have a natural tendency to flow in all directions, whereas the treble frequencies tend to "beam" in the direction that the speaker faces. Hence concentration of the sound within a 90° angle is most beneficial with respect to the low frequencies. There are also other, more technical reasons why a corner location favors bass reproduction.

If your audio system is stereo, for fullest bass response the two speakers must be in phase. This means that when the same signal

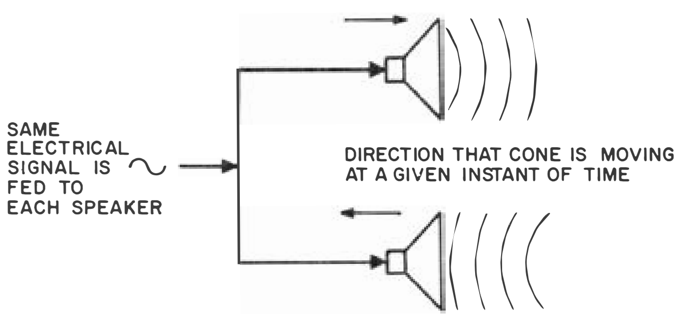


Fig. 604. *Speakers operating out of phase.*

appears in both channels, the cones of both speakers should move out together and in together, as illustrated in Fig. 603. If the two speakers are out of phase—the cone of one speaker moving in when the cone of the other is moving out, as depicted in Fig. 604—a certain amount of sound cancellation takes place, especially at the bass frequencies.

Testing for correct speaker phase is relatively simple in most cases. First, place the stereo speakers as close together as feasible. Then feed a low-frequency signal into both channels of the stereo amplifier. The source may be a mono test record or test tape, or (Fig. 605) it can be yourself—touch your finger to the hot lead of

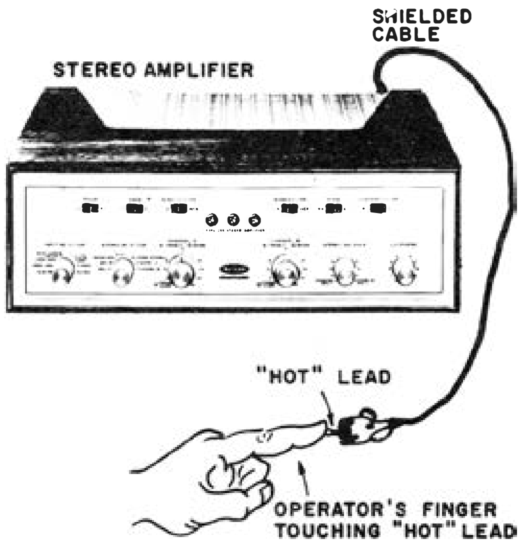


Fig. 605. *Feeding a 60-cycle signal into a stereo preamplifier to phase the speakers. See also Fig. 202 on page 17.*

a shielded cable plugged into a high-level input jack of your stereo amplifier. Set the selector switch of the preamplifier so that it admits the low-frequency signal. Set the function switch of the preamplifier so that the signal goes to both channels and hence to both speakers. Have someone, possibly yourself, stand in a position in front of and equidistant from the two speakers. If the speakers are in phase, the sound coming from them will be maximum. But if the speakers are out of phase, the sound will be diminished. If your stereo amplifier has a phase-reversal switch, flip this back and forth to see which position of the switch produces the most sound. If maximum sound occurs when the switch is in the in-phase position, nothing more need be done. If maximum sound occurs when the switch is in the out-of-phase position, reverse the leads to *one* of the speakers. Then check again. When you have identified the correct connections of the leads to the speaker terminals, mark the leads and terminals in recognizable fashion, for example, with fingernail polish.

If your stereo amplifier lacks a phase-reversal switch, have an assistant reverse the connections at the speaker terminals; have him merely touch the leads to the terminals, so that reversal can be quickly accomplished. Or, preferably, you can fashion a lead-reversing device from a double-pole double-throw knife switch, as in Fig. 606.

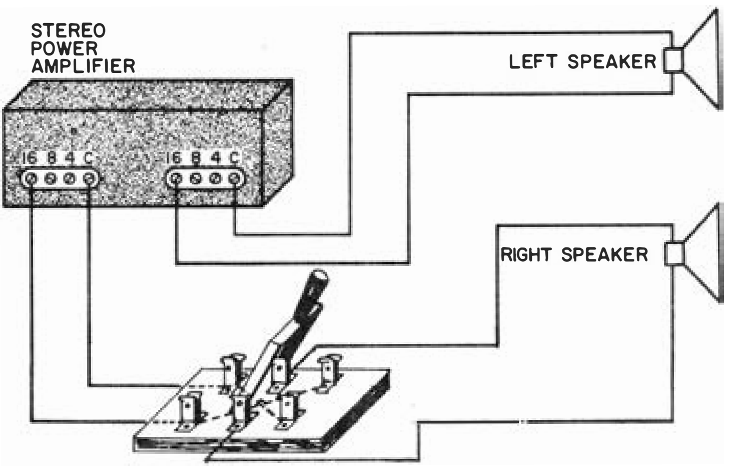


Fig. 606. *Use of a double-pole double-throw switch to reverse the polarity of the stereo speakers.*

Another phasing method involves use of the RCA phase checker. This consists of two small speakers employed as microphones and therefore called receptors. The output of the receptors goes to either a sensitive voltmeter or an oscilloscope, which means that you must have access to one of these items. If you do, then one receptor is placed against each speaker as you feed the same low-frequency signal to both speakers. One receptor contains a two-position switch marked in-phase and out-of-phase. Flip the switch back and forth. If the meter or oscilloscope shows maximum signal when the switch is in the in-phase position, your speakers are correctly phased. If not, reverse the leads to one of the stereo speakers.

Amplifier problems

It seems almost trivial to say that your bass control may be set too high or too low. But since experience shows that this kind of thing does happen, it must be mentioned. In a dim light, and with poorly marked tone controls, it is not always clear where these controls are set. A child or a visitor may have turned them to an extreme position.

Most speakers can use some bass boost, even at high levels, so don't be afraid to use at least a moderate amount of bass boost. Too many persons are psychologically strapped to the "flat" position of the tone controls. Keep in mind that flat electrical response does not necessarily mean flat acoustic response.

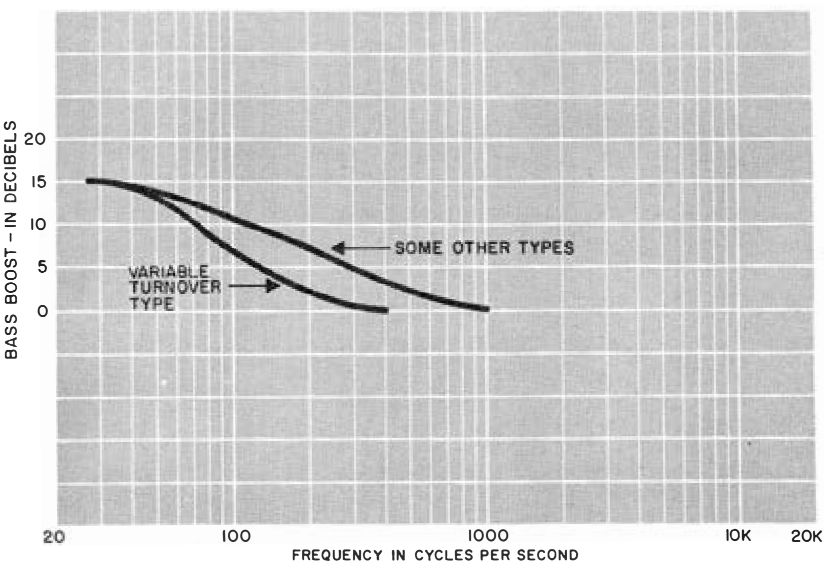


Fig. 607. *Comparison of bass boost typically provided by Baxandall type circuit with that provided by some other circuits.*

High-fidelity speakers generally don't fall off seriously in bass response until below 60 or 70 cycles. Then they can use assistance from the bass control. Unfortunately, to provide substantial assistance below 70 cycles or so, the bass control must commence the rise in response above 70 cycles. The result may be a boomy sound. The likelihood of this depends in part upon the design of the bass control. This control preferably utilizes what is known as a variable turnover circuit, which tends to confine the bass boost more nearly to the area where it is really needed. Fig. 607 helps make this clear. It shows two bass-boost curves, both resulting in the same maximum boost, but one of them beginning to provide bass boost at a substantially higher frequency than the other. The curve that covers a larger frequency range is apt to be the less desirable one, because it may well supply boost where it isn't needed. In shopping for a preamplifier, score a point or two for the ones that claim to use the variable turnover circuit.

If the treble control is set too high, this by contrast makes the bass appear insufficient. Conversely, too low a setting will appear to augment the bass. (In inexpensive radios and other audio equipment, the so-called bass control is often nothing more than a device to reduce the treble.)

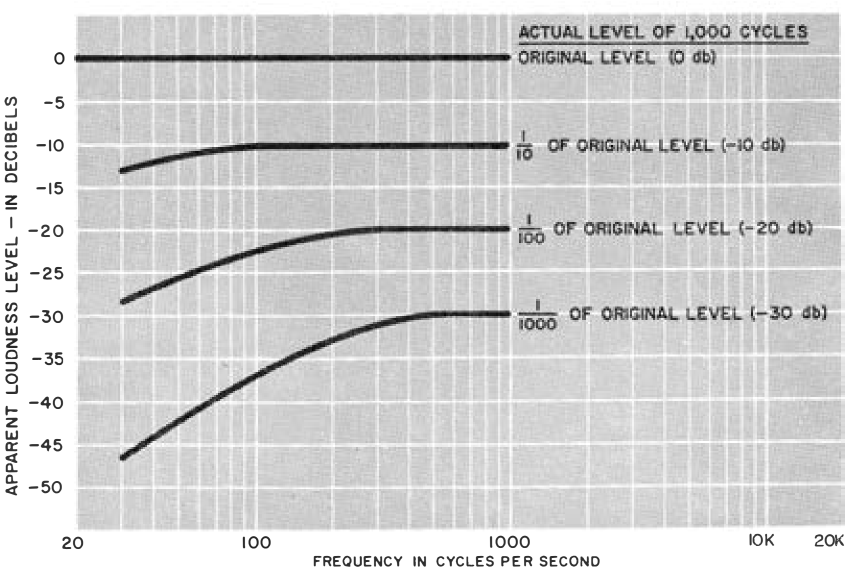


Fig. 608. *Decline in apparent loudness of frequencies below 1,000 cycles when 1,000 cycles is reproduced 10, 20 or 30 db below the original sound level.*

What is known as the Fletcher-Munson effect means that the bass and treble frequencies, especially the bass, seem to fade out more rapidly than the mid-range when volume is reduced below the level of the original music or speech. This effect is portrayed in Fig. 608. Most preamplifiers attempt to compensate for this characteristic of the human ear by automatically supplying large amounts of bass boost (and sometimes moderate amounts of treble boost as well) when you turn down the volume control to less than realistic levels. All you have to do to get this automatic boost is to flick the "loudness" switch, which converts the volume control into a "loudness" control. Sometimes the switch has more than one position, providing varying degrees of bass (and treble) boost.

However, I personally feel that the loudness control introduces more problems than it solves, and therefore suggest that you try using the bass control instead of the loudness switch to compensate for the Fletcher-Munson effect. If you prefer to use the loudness

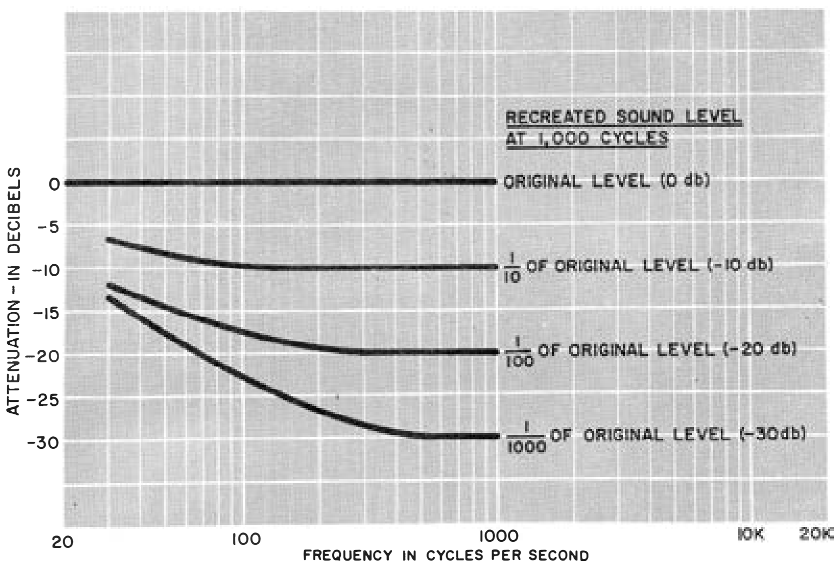


Fig. 609. *Action of loudness control (as volume at 1,000 cycles is reduced below the original level, there is less attenuation of frequencies below 1,000 cycles).*

switch, understand the problems it entails so that you may use it wisely toward maximum listening pleasure.

In the early 1930's, Messrs. Fletcher and Munson presented data on the *average* results when a group of persons were tested with respect to the apparent disappearance of bass as volume is reduced. The loudness control provides bass boost which is based on these *average* hearing characteristics. But as individuals, most of us depart from the average, some of us considerably so. Accordingly, a loudness control may provide too much bass boost for one person and too little for another. It seems logical to rely on the bass control, which enables an individual to tailor the bass boost to his specific requirements.

Another problem inherent in the loudness control is that it can operate correctly only if its maximum position corresponds to the original sound level, for example as you might have heard it in a concert hall. Then as you back down on the control, it introduces bass boost to compensate for the seeming disappearance of bass as volume is reduced below the original level. The more you back down on the control, the greater is the amount of bass boost relative to the rest of the audio range (Fig. 609).

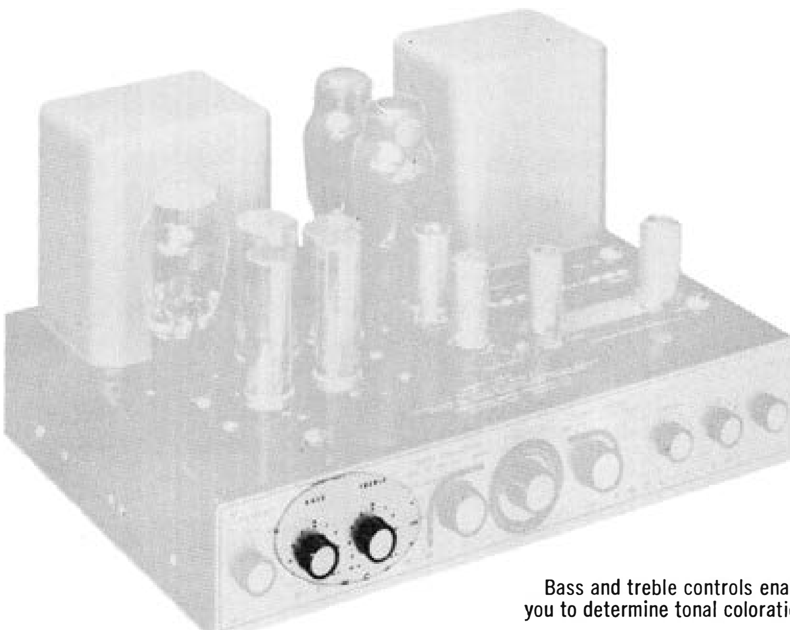
Thus you have the problem of seeing to it that no matter what your signal source may be—tuner, phono, tape recorder or TV—the maximum position of the loudness control should always result in your hearing the same volume of sound as at the original performance. If the maximum position of the control results in your hearing a sound level greater than what you would have heard at the original performance (which is very often the case because most of us like to have an ample reserve of sound volume), the loudness control tends to introduce excessive bass. Hence the user of the loudness control frequently finds that, instead of merely filling in the missing bass, it exaggerates the bass to the detriment of listening pleasure.

Many preamplifiers contain a rumble filter to eliminate turntable rumble and other noise in the very-low-frequency range, such as that produced by friction between the pressure pads and the tape in a tape machine. An ideal rumble filter should attenuate very sharply and only below 50 cycles or so. Unfortunately, it is rather difficult and expensive to build such a filter. Therefore, the filter frequently sweeps away not only the rumble frequencies but also a substantial portion of the desired bass frequencies. If you find that bass is deficient, it may be that you have inadvertently left the rumble filter switch in the on position. If rumble is so bad that you find you can't do without the filter, the real answer to the problem lies in replacing the component that produces such excessive rumble.

Tuner problems

Some FM stations are prone to emphasizing the bass region, while others aim for a brilliant effect by diminishing the bass and emphasizing the treble. This is why you have tone controls in your audio equipment. It is for *you* to decide, on the basis of your experience with live music, whether bass should be augmented, diminished or left alone, to achieve the most pleasurable reproduction. The tone controls more or less enable you to cancel such tonal coloration as the broadcaster has supplied.

In the preceding chapter on noise, it was stated that most FM tuners have afc (automatic frequency control) to keep the tuner from drifting off station and thereby producing noise. If the afc circuit is poorly designed, or if a certain part (a capacitor) in it goes bad, this circuit may cause appreciable bass attenuation.



Bass and treble controls enable you to determine tonal coloration.

A simple test enables you to determine whether the afc circuit is defective with respect to bass reproduction. Most FM tuners incorporate a switch that permits you to eliminate the afc action. With afc off, tune in a program that has substantial bass content. Put the afc switch in the on position. If you notice a reduction in bass, the afc circuit is defective. A qualified technician can easily and speedily cure the trouble, generally by replacing the filter capacitor in the afc circuit with a unit of appropriately higher value.

Phono problems

In the case of the magnetic phono cartridge (and also the ceramic pickup if used with an adapter so that it may be fed into the magnetic phono input jack of a preamplifier), improper bass response is mainly apt to be the fault of the equalization circuit in the preamp. The preamp is supposed to furnish a specific amount of bass boost, known as the RIAA curve (which also calls for a specific amount of treble cut), as shown in Fig. 610. In some preamps there is a tendency to provide less than the required amount of bass

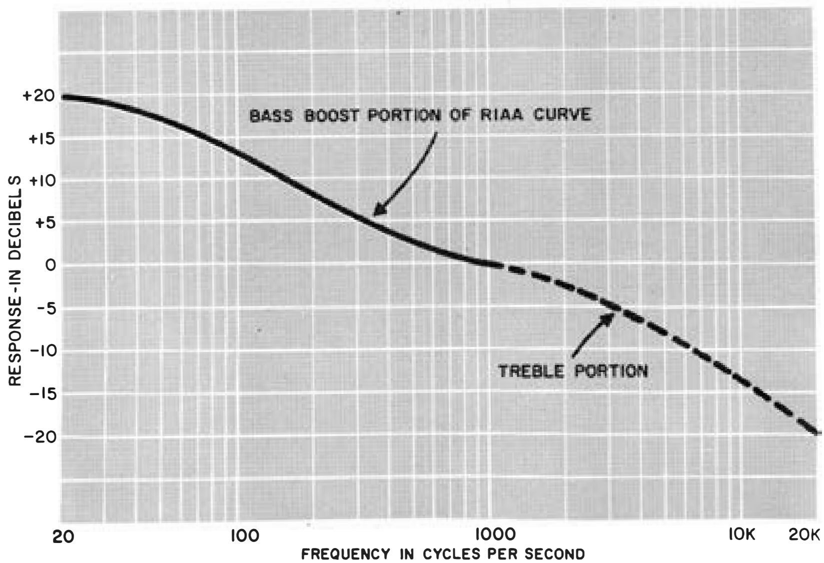


Fig. 610. *Bass boost called for by RIAA phono playback curve.*

boost as a means of overcoming the problem of hum. If you suspect that your preamp is skimpy on phono bass boost, a service technician can check your suspicions in very short time.

Improper bass may be due to pure carelessness on your part. Most preamps provide not only RIAA equalization, which has been virtually standard since 1954, but also several other equalization curves appropriate to phono discs of earlier years. These generally involve less bass boost than does the RIAA curve. If you aren't getting enough bass, it may simply be because the equalization switch is set to the wrong equalization curve. Or, when playing old discs, you may be getting a tubby sort of bass because you have the equalization switch set to the RIAA curve.

This switch often has a position that provides the bass boost which is needed when feeding a tape-head signal directly into the preamp. Such boost is substantially greater than RIAA boost. Accordingly, if you play a phono disc but have the equalization switch set to the tape-head position, bass will be greatly exaggerated.

Proper bass response becomes more troublesome when a ceramic pickup is used without an adapter and therefore is fed into a high-level input jack of the preamplifier. The problem here is that the bass response of the ceramic cartridge depends very much on the "load" resistor placed across it. In every preamp there is a resistor across every input jack. At a high-level jack, the value is generally around 510,000 ohms. However, the typical ceramic pickup requires a load resistor of about 2 or 3 million ohms to yield adequate bass output. A smaller resistor, such as 510,000 ohms, results in considerable bass loss.

One cure is to replace the 510,000-ohm load resistor in the preamp with one having the value called for by the cartridge manufac-

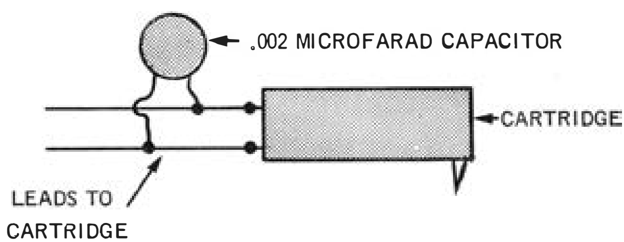


Fig. 611. *Wiring a capacitor across a ceramic cartridge to improve its bass response.*

turer. A perhaps simpler cure, which doesn't require going inside the preamp, is to wire a capacitor directly across the leads to the ceramic cartridge, as in Fig. 611. Usually a value around .002 microfarads works out satisfactorily. If bass still isn't sufficient, one might go as high as .003 microfarads. (On the other hand, a capacitor employed in this manner also attenuates the signal level; therefore you may have to be careful not to use too large a capacitor.) A miniature capacitor can be used, soldered directly across the connecting leads in the shell containing the cartridge. If you aren't at home with a soldering iron, simply remove the shell from the tone arm, take it (including the cartridge of course) to an audio technician, and ask him to solder a .002 microfarad capacitor across the leads. It takes but a few moments.

Tape recorder problems

The bass reproduction problems of the tape recorder are substantially similar to those of the magnetic phono cartridge. Equalization may be wrong in the tape recorder or, if you are feeding the playback-head signal directly into the audio preamp, equalization may be incorrect in the preamp. These are things to be checked, and corrected if necessary, by a technician.

Some tape machines incorporate tone controls, and it is possible that improper bass response is due to incorrect setting. Consult your instructions for the proper position. If they do not clarify the matter, or if bass response seems poor despite your having followed the instructions, you can do this: Make a tape recording of a phono disc having substantial bass. Play back the tape and synchronize it with the phono disc; use the selector switch of your preamp to alternate between the two. Find the position of the tone control that makes the tape playback sound as close as possible to the phono disc.

Most tape machines are designed to operate at two or more speeds. In the majority of machines, equalization is automatically changed as you go from one speed to another. In a few cases, however, the change in equalization must be made manually. Incorrect bass response may result from your having forgotten to make the necessary change in equalization in recording or playback or both.

Fig. 612. *Rotating a tape playback head about its vertical axis for possible improvement in bass response.*

Sometimes bass is excessive because of certain characteristics of the tape playback head. What happens is that, in the low-frequency range, the head exhibits a series of dips and peaks in response, with the peaks exceeding the dips in magnitude. Sometimes these can be smoothed out by slightly rotating the head about its vertical axis, as in Fig. 612, so that the tape approaches it at a somewhat different angle. The feasibility of rotating the head in this manner depends upon the design of your tape machine. Ordinarily, such an expedient should be left to the technician.

Chapter 7
Treble Problems

HIGH FIDELITY IS CLOSELY ASSOCIATED WITH THE ABILITY TO REPRO-duce sounds to at least 15,000 cycles and preferably beyond. While the average adult can hear little if anything above 15,000 cycles, response in this region is nevertheless considered necessary because it is equivalent to the ability to reproduce the abrupt starts and stops of natural sounds.

Some components, in the endeavor to extend response to at least 15,000 cycles, produce an undesirable side effect—a peak in treble response in the vicinity of 15,000 cycles or lower. Such a peak produces a strident or shrill effect. Therefore the objective of high fidelity is to extend response *smoothly* to 15,000 cycles or higher.

Speaker problems

As described in the preceding chapter, a number of speaker systems have a balance control that varies the tweeter level relative to the woofer. Excessive or insufficient treble may be due to an inappropriate setting of this control.

In a heavily draped room, with much soft furniture and with deep pile carpeting, there is substantially greater absorption of high frequencies than in a modernistic room where the sound bounces off hard, reflecting surfaces. Hence your choice of a speaker may be governed by the room in which you plan to put the speaker. Or, having selected a speaker without consideration of room characteristics, you will want to adjust the balance control (if there is one) according to the room. Consult your own ear as to correct woofer–tweeter balance; try for what seems to you to be the most natural sound. You may want to experiment for several days or longer with various positions of the level control. After you have decided which position seems to achieve best balance overall, leave the control alone.

The position of the speaker with respect to both the room and the listener can affect apparent treble response. If the speaker points

It may take several days of experimentation for you to find the control settings that supply the "most natural" sound.

directly at the listener, the tendency of the treble notes to "beam" straight ahead may give the impression of overmuch treble. Conversely, if the speaker points away from the listener, particularly if it faces sound-absorbing material such as an overstuffed couch, treble may appear smothered. An intervening object between the speaker and the listener may cause treble loss, possibly a desirable loss.

It may be feasible, particularly in a home-assembled speaker system, to vary only the position of the tweeter. Thus if the treble seems overbright, one might allow the tweeter to "splash" off a wall instead of aiming at the listener. A similar effect might be gained by pointing the tweeter upward. In fact, there are a few commercial speaker systems where the speaker faces upward instead of to the front.

One attribute of stereo is directionality. Sounds appear to come from left, center, right and intermediate locations. Directionality is more dependent on high frequencies than on low ones. Therefore, while it is advisable in general to have matched speakers for stereo, this is especially advisable in the case of the tweeters. It is better to have identical tweeters and different woofers than the other way around. If the tweeters are mismatched, sounds tend to jump from one speaker to the other. Thus a vocalist may appear to be at the left one moment and at the right the next, depending on the frequency of the note being sung.

Amplifier problems

Dissatisfaction with treble response often stems from improper setting of the treble control. The proper setting depends upon at least four factors:

1. Frequency balance of the signal source—phono disc, prerecorded tape, or radio broadcast (including TV).
2. Volume level at which you are listening. At low levels, it may be necessary to supply a little treble boost because treble frequencies *seem* to fall off more rapidly than mid-frequencies as volume is reduced. (On the other hand, some loudness controls automatically supply moderate treble boost at reduced volume levels.)
3. The acoustic nature of the room in which you are listening. Some rooms "soak up" the treble, while others appear to accentuate it.
4. Your personal listening characteristics and tastes. Perhaps you care for treble appreciably more or appreciably less than the average person. Perhaps your hearing acuity above 10,000 cycles has diminished and you wish to compensate for the loss.

The upshot of these various considerations is that you should adjust the treble control so that the reproduced program material sounds the most natural and pleasing to *your* ear. Of course, if others are going to do the listening, encourage them to adjust the treble control to their taste. Merely leaving the treble control in mid-position, where response is electrically flat but not necessarily acoustically flat, is hardly the way to achieve high fidelity.

Mid-setting of the treble control may not even give you electrically flat response. In high-quality preamplifiers, there is seldom a serious deviation from flat response when the treble control is in the 12 o'clock position. But in components of lesser quality,

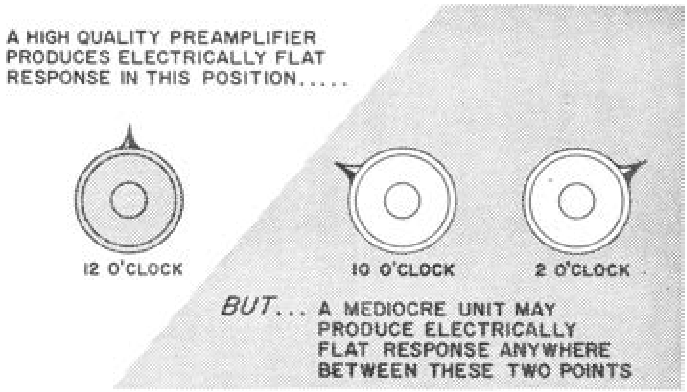


Fig. 701. *Various treble control positions that may produce electrically flat response.*

serious deviations are found from time to time. Hence one may find that flat response requires the control to be set perhaps as low as

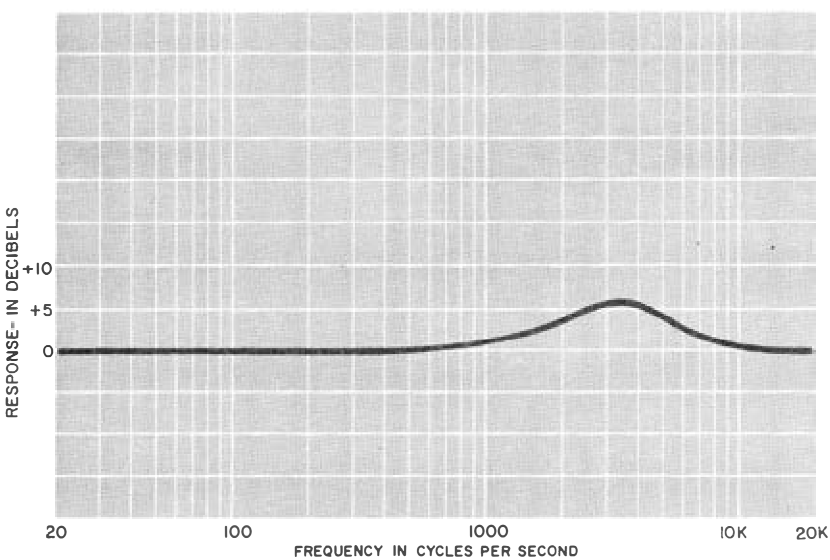


Fig. 702. *Typical effect of a presence control on frequency response.*

10 o'clock or perhaps as high as 2 o'clock, as illustrated in Fig. 701.

In a number of preamplifiers, treble response tends to vary with the setting of the volume control. Whereas response may be quite flat to 15,000 cycles (or more) at very low and at very high volume settings, it may drop appreciably above 10,000 cycles at interme-

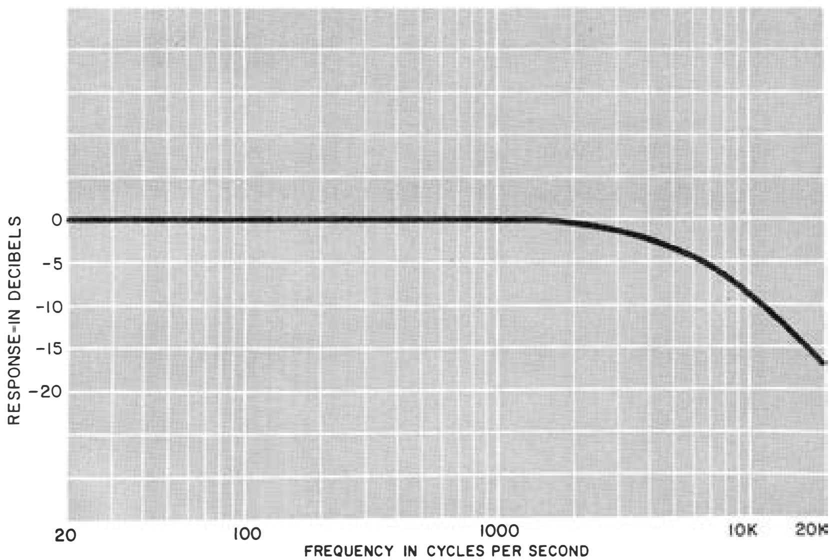


Fig. 703. *Typical effect of a treble filter on frequency response.*

diate settings, where most listening is done. This is a problem of design, one that not every design engineer copes with adequately. Therefore it is all the more incumbent upon you to adjust the treble control until response sounds correct to your ear.

Some preamps contain a presence control, which at the flick of a switch gives about 6-db emphasis to frequencies around 3,000 to 5,000 cycles, as shown in Fig. 702 (6-db represents a quadrupling of acoustic energy). The effect is to "crispen" the sound. Thus it may give the illusion of bringing a soloist forward of the orchestra. Judicious use of the presence control can augment listening pleasure. But don't use this control indiscriminately. Certain speaker systems are deliberately designed to accentuate response in the 3,000–5,000-cycle region. If speaker emphasis is coupled with additional emphasis by the presence control, the total effect may well be too much of what was previously a good thing. Similarly, some program material on disc, tape or air may have a deliberate boost in the mid-treble region, making it unwise to use the presence control.

Most preamplifiers incorporate a treble filter to do away with much of the record noise on discs, particularly the 78-rpm discs of another era. Of course, this switch can also be used to cope with a particularly noisy tape or radio broadcast. But, as shown in Fig. 703, the treble filter cuts out desired audio frequencies as well as undesired noise. The theory is that, for a noisy source, the filter cuts out more noise than program material, resulting in a net improvement. But if you inadvertently have the treble filter in the on position when listening to relatively noise-free material, you are probably robbing yourself of a certain amount of listening pleasure.

Tuner problems

An FM broadcast station is always required to supply a specific amount of treble boost (Fig. 704). An FM tuner should supply a corresponding quantity of treble cut, as shown in Fig. 704, thereby restoring flat response. This equalization procedure aims at improving the signal-to-noise ratio. The amount of equalization is technically designated by the term "75 microseconds de-emphasis" or by the term "2,122-cycle turnover."

It has been a practice of some tuner manufacturers to introduce too little de-emphasis, resulting in exaggerated treble. The added brilliance, it is thought, will persuade some buyers that the tuner in question has higher fidelity than other tuners which adhere faithfully to the standard de-emphasis requirement. On the other hand, some tuners provide too much treble cut, resulting in a dull sound.

If your FM tuner appears to have excessive or insufficient treble response, it requires an audio technician but a few moments to check it out in this respect. If correction is required, all that is usually necessary is replacement of one inexpensive resistor or capacitor.

Poor treble response on the part of a stereo FM tuner will affect stereo separation (distinction between left and right). For good separation the tuner must have virtually flat response to about 50,000 cycles. If you are feeding the signal from a conventional FM tuner into a multiplex adapter, you must make sure that the signal is taken at a point *prior* to the tuner's de-emphasis network.

If there is any doubt, check with a technician. If the signal is taken after the de-emphasis network, there will be great treble loss long before 50,000 cycles, and stereo separation will be virtually nil. If you are going to do all your listening through the multiplex adapter, it is a good idea to have the technician disable the tuner's de-emphasis network, which merely means snipping a resistor or capacitor lead.

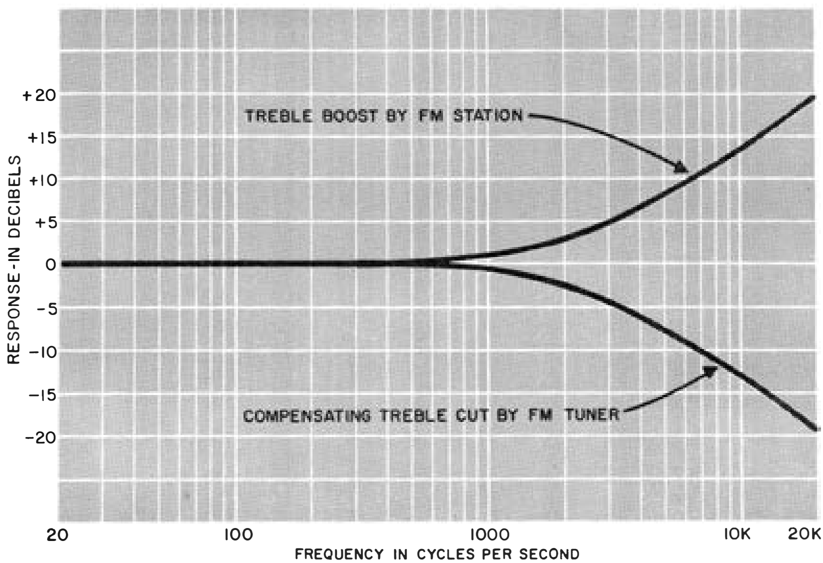


Fig. 704. *Equalization in FM broadcasting and reception.*

In FM tuners of the pre-multiplex era, there will quite possibly be a substantial drop in treble response even prior to the de-emphasis circuit. Sometimes a technician can make simple changes —such as shortening certain leads—that will keep response reasonably flat to 50,000 cycles. In other instances, the only course open to you, if you wish good stereo separation, is to purchase a new FM tuner.

Phono problems

Across the magnetic phono input jack of the preamplifier is a "load resistor." Its value has an important bearing on the treble response obtained from the magnetic cartridge you are using. Each brand of cartridge has its own optimum value of load resistance,

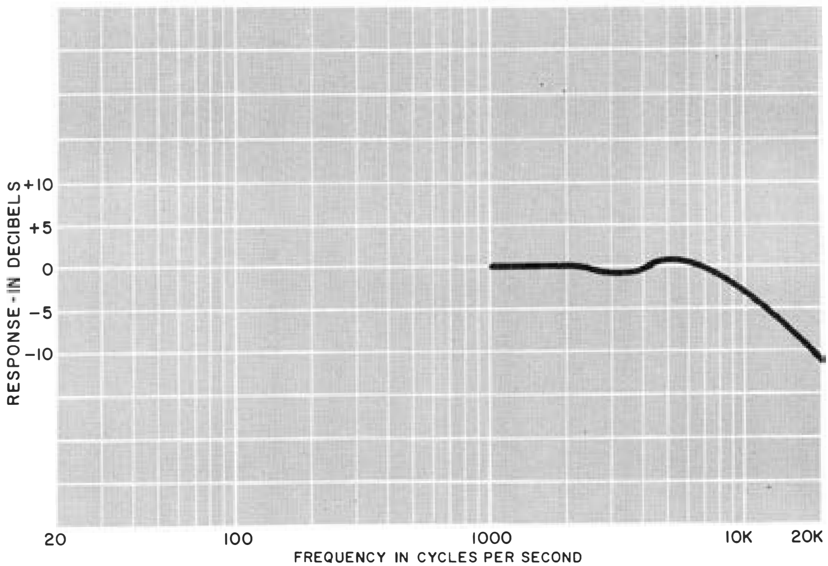


Fig. 705. *Treble loss that might result from too small a load resistor for the magnetic pickup.*

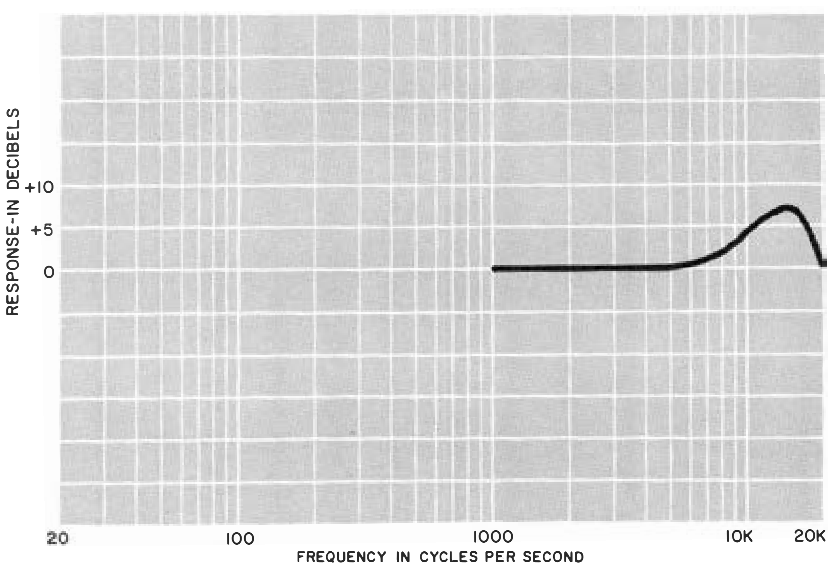


Fig. 706. *Treble peak that might result from too large a load resistor for the magnetic pickup.*

usually between 47,000 and 100,000 ohms. It is important that you ascertain whether the preamplifier of your choice provides the load resistance specified by the manufacturer of your cartridge. If the resistor has to be changed, this is a simple and inexpensive matter, one that you can probably attend to yourself if you have had kit-building or similar experience.

Too small a load resistor across a magnetic cartridge will produce treble cut, as illustrated in Fig. 705. If the resistor is too large in value, there will be a peak in treble response (Fig. 706).

Too much or too little treble may be due to a fault in the equalization circuit contained in the phono section of the preamp. This circuit should produce the specific amount of treble cut called for

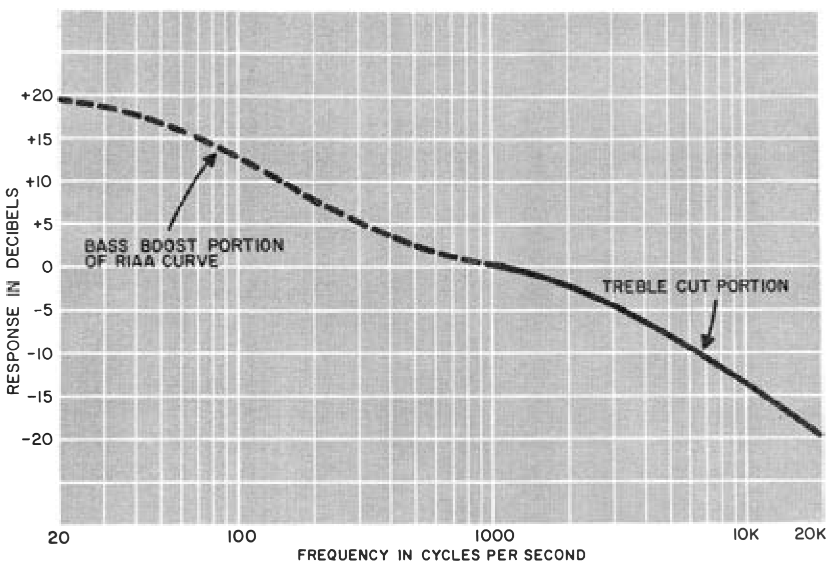


Fig. 707. *RIAA phono playback equalization.*

by the RIAA curve (Fig. 707). The circuit is a relatively simple one, and accurate treble cut is essentially a matter of how much care the preamp manufacturer has taken to use resistors and capacitors that are (1) close to design value and (2) of a quality that resists significant change in value with age and use.

A ceramic cartridge can be employed by feeding it *directly* (without an adapter) into a high-level input jack. When used in this manner, the cartridge is "self-equalizing" in the treble region.

This means that the cartridge has a certain amount of treble boost built-in by the manufacturer. (Note that the ceramic cartridge by its physical nature responds to the phono disc in such a manner that treble *boost* is needed, whereas the magnetic cartridge, in contrast, needs treble *cut.*) However, depending upon the individual cartridge, this built-in boost may be too much or too little.

By interposing a special adapter between the ceramic cartridge and the phono amplifier, the ceramic pickup can be converted into the equivalent of a magnetic cartridge, which means that it now requires the same equalization as a magnetic. The problem then arises: What to do with the built-in treble boost of the ceramic pickup, which is no longer needed? One function of the adapter is to cancel the cartridge's treble boost. If you buy the adapter from the manufacturer of the ceramic cartridge you are using, you maximize the chance of correctly canceling the built-in treble boost.

Not infrequently, an audio salesman will suggest that the selection of the phono cartridge be geared to the selection of the speaker. Undesirable as it may be, every speaker system has sound coloration. It is often feasible to balance out some of this coloration by using a cartridge with opposite characteristics. For example, if the speaker is on the brilliant side, you may do well in choosing a phono cartridge somewhat on the dull side.

Many phono cartridges employ a small bit of rubber or other "damping" material to restrain the movement of the stylus. The resiliency of this material affects the treble response of the cartridge. With age, the damping agent may dry out and become hard, resulting in a loss of response at the very high end of the audio range. If your cartridge has apparently lost its original sheen and brilliance, it may be time to replace the stylus assembly even though the stylus itself has not worn sufficiently for replacement.

Tape recorder problems

Both in recording and playback, equalization must vary with tape speed to achieve a flat response or as close an approximation as is feasible. In most home machines, the change in equalization is automatic with change in speed. However, some units require that the equalization change be made manually by moving a knob or switch. Unless you are careful, this can give rise to a tape recorded with incorrect equalization. This is a permanent error, unless you are in a position to remake the tape. On the other hand, an error in playback equalization is always subject to immediate correction.

In recording, a high-frequency current, typically between 50,000 and 100,000 cycles, is fed to the record head along with the audio signal to minimize distortion and increase the amount of signal recorded on the tape. This is called bias current. Unfortunately, a slight excess of bias can result in substantial treble loss. Too little bias will accentuate treble. Checking and adjusting bias current is ordinarily a matter for the audio technician.

Still, if your tape recorder has separate record and playback heads, permitting simultaneous recording and playback, you might try the following technique of adjusting bias current, provided that the machine has a bias control (usually accessible internally). Record a high-quality phono disc and simultaneously play back the tape. Adjust bias current so that the frequency response of the tape is as close as you can get to that of the phono disc. At the same time, watch out for an undue increase in distortion. If distortion goes up, you may have to accept a slight amount of treble loss to avoid a significant increase in distortion.

For response to 15,000 cycles at 7.5 ips, the gap of the *playback* head must be no wider than .00025 inch. For response to 10,000 cycles or so at 3.75 ips, the gap has to be of the order of .00015 inch. Modern-day playback heads frequently have gaps in the vicinity of .0001 or .00012 inch. However, the gap tends to widen gradually with use—sometimes fairly rapidly in the case of a cheap head—resulting in diminution of treble response. Head replacement is then called for. Since good tape playback heads are quite expensive, it is probably advisable to have a technician check out the playback response of your tape machine before resorting to a new head.

Very minute separation between the tape and the heads can produce appreciable treble loss, particularly in playback. Separation may be due to dirt or tape oxide on the heads. Therefore the heads should be cleaned after about every 8 hours of use. Ordinarily you can use a cotton-tipped stick dipped in alcohol. But to be on the safe side, consult the instructions that come with your tape recorder as to what cleaning agent to use. Avoid carbon tetrachloride ordinarily, because it may eat into the nonmetallic portion of the head; also, its fumes are poisonous.

Poor tape-to-head contact may be due to a faulty pressure pad; that is, the tension adjustment of the pad. This adjustment is perhaps best left to the technician because excessive pressure may result in a considerable increase in wow and flutter, and perhaps cause tape squeal.

The gap of the head should be exactly at right angles to the long dimension of the tape (Fig. 517, p. 61). This is termed azimuth alignment. If the same head is used for recording and playback, a slight departure from correct azimuth hardly matters. But if the machine employs separate record and playback heads, and if the two do not have the same azimuth, even a slight difference will cause substantial treble loss. Similarly, if you play a prerecorded tape, you will have appreciable treble loss if your playback head deviates from correct azimuth. Hence it is desirable from time to time, perhaps once or twice a year, to check azimuth by means of an azimuth alignment tape, available at audio stores. Play the test

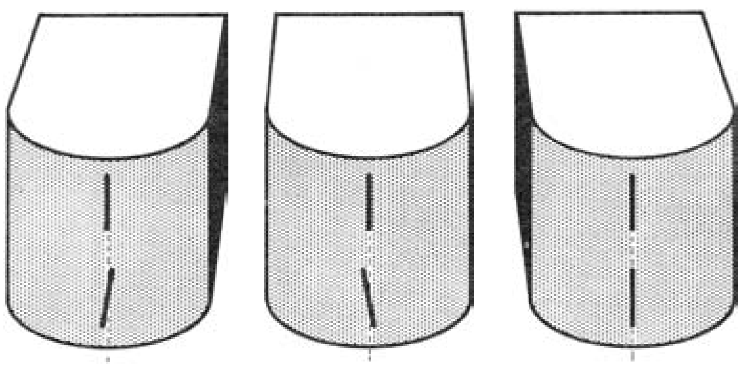


Fig. 708. *Colinearity of a stereo tape head.*

tape and adjust the playback head for maximum output. If the machine has a separate record head, it is then necessary to align the latter by feeding a high-frequency signal into the tape recorder (from a test disc or audio oscillator) and adjusting the record head for maximum output while simultaneously recording and playing a fresh tape.

A stereo head, containing two gaps, frequently offers a special azimuth problem. Theoretically, the two gaps should be exactly colinear, that is, exactly in the same vertical line. In practice they are sometimes slightly out of line, as in Fig. 708. Therefore it is impossible to obtain maximum treble response simultaneously on both stereo channels. Instead, it is necessary to adjust the head to a compromise position such that treble response is about the same on each channel.

A great amount of treble boost is required in tape recording to overcome certain magnetic phenomena. This boost tends to overload the tape at high frequencies. Fortunately, the frequency content of most sounds is substantially less in the treble region than in the mid-range (Fig. 709). Hence the natural drop in treble *tends* to compensate for the accentuation of treble by the tape recorder. This compensation is not always complete, resulting in

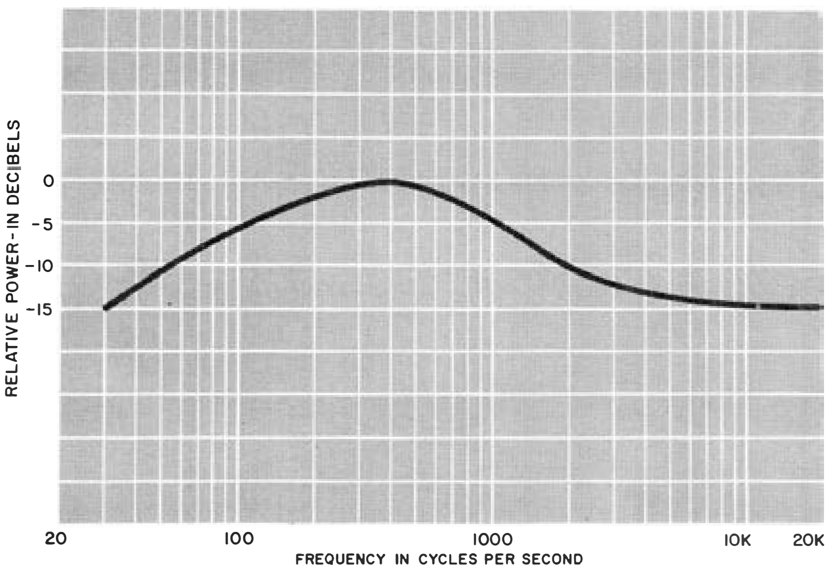


Fig. 709. *Typical variation of acoustic power with frequency for orchestral music.*

some overloading of the tape at the high end. The higher the level at which you record, the greater is the likelihood of such overloading. Once the tape is saturated (completely overloaded), no matter how much more signal you *apply* to the tape, you cannot increase the amount of signal recorded on the tape. When saturation occurs first at the high frequencies, the net result is to attenuate these frequencies relative to the rest. In brief, a high recording level may cause treble loss (and also considerable distortion, as discussed in the next chapter).

The practical upper limit of frequency response depends upon tape speed. As this is written, it is practical in home tape recorders to extend response smoothly to 20,000 cycles or higher at 7.5 ips, at the same time maintaining an excellent signal-to-noise ratio and low distortion. At 3.75 and 1.875 ips, response can be maintained to about 16,000 cycles and 13,000 cycles, respectively, although with some sacrifice in terms of noise and distortion. In fact, you could achieve treble response at 3.75 and 1.875 ips virtually as good as at 7.5 ips, but only by making further sacrifices in terms of noise and distortion. The moral is that it may be premature as yet to look to speeds below 7.5 ips for excellent treble response if the other characteristics of high fidelity (such as low noise and distortion) are also important to you. In fact, if you seek the ultimate in home performance, you might want a machine that operates at 15 ips, not so much for still better treble response, but for substantially the same kind of response as at 7.5 ips together with lower noise, lower distortion and lower wow and flutter.

Heads tend to become gradually magnetized with use. This results in some loss of treble response because a magnetized head tends to erase, particularly at high frequencies. Therefore heads should be demagnetized after about every 8 hours of use. If a head has been exposed to a very sharp, loud sound—for example, if someone has knocked over the microphone while recording—the head should be demagnetized as soon as possible.

High-frequency response will vary somewhat among brands of tape and among types of tape (1½-, 1- and ½-mil tape). Accordingly, a problem of too little or too much treble can be aided by proper choice of tape. (At the same time, remember that there are a number of factors in addition to treble response to be considered in choosing a tape—thickness of the base, freedom from squeal, constancy of output level, resistance to flaking, immunity to print-through, etc.)

Cable problems

Shielded cables are a threat to treble response throughout the audio system. The cable has an electrical characteristic called capacitance, which acts as a partial short circuit, particularly as frequency rises. Therefore these two rules should be obeyed in running shielded cable between any two components: (1) Keep the cable as short as practicable. (2) Use cable of low capacitance per foot; this means cable of relatively large diameter as opposed to the thin shielded cable that one sometimes finds in use. A standard "microphone" cable, attractive in appearance and not too bulky and having a capacitance of about 25 to 30 picofarads (pf) per foot, is readily available and widely employed.

In audio components of good quality, design precautions are taken against the effect of cable capacitance. Thus the typical high-quality preamplifier has a "cathode-follower" or "anode-follower" output, permitting a substantial run of cable to the power amplifier without loss of treble. Other components, such as tuners and tape recorders, also often contain a cathode- or anode-follower output. But to be on the safe side, it is nevertheless advisable to keep the cable short and of low capacitance per foot (Fig. 710).

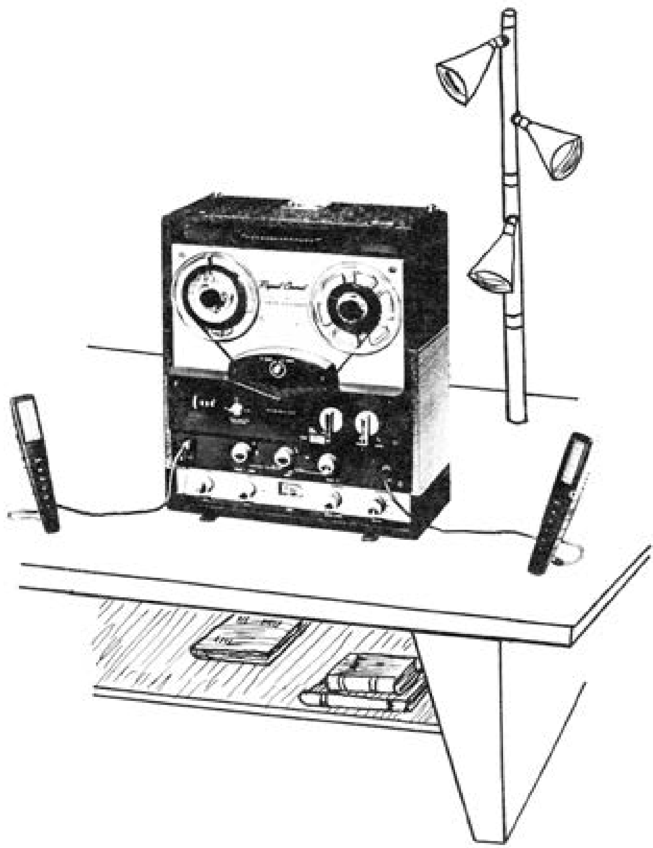


Fig. 710. *Minimizing the capacitance of the microphone cable.*

This precaution is especially important in the cable that connects the magnetic phono pickup or the tape playback head to the following component. Here a cable of high capacitance can seriously impair treble response.

Again of special importance is the cable between the microphone and tape recorder if the microphone is of the magnetic (usually dynamic) type. Most home tape machines are designed to accept the signal from a high-impedance microphone, and in this case you are in danger of appreciable treble loss if the cable is much over 12 feet long or so. If you must have a long run of cable, use a low-impedance microphone, which permits a length of 50 feet or more. At the same time, a low-impedance microphone requires that you have a stepup transformer between the far end of the cable and the tape recorder. (In a very few cases, such a transformer may already be built into the tape recorder.) On the other hand, if you are using a piezoelectric (crystal or ceramic) microphone, all frequencies are equally attenuated by cable length, that is, the cable does not attenuate the treble frequencies relative to the rest of the audio range.

Chapter 8
Distortion

IN SUCCESSIVELY LARGER AMOUNTS, DISTORTION IMPARTS A CLOUDY, grainy, fuzzy, gritty, coarse, broken or completely unintelligible aspect to reproduced sound. In small amounts, distortion may not be immediately evident but becomes apparent after a while as "listening fatigue"; the sound has an undefinable quality which makes you want to shut it off. But when distortion is kept very small, the sound has an effortless, transparent quality that invites protracted listening.

The kind of distortion that bothers us most consists of unwanted signals added to the desired audio signals. These unwanted signals, known as distortion products, are the result of imperfections in the audio equipment. The most irritating distortion products are harmonic and intermodulation. Harmonic products (Fig. 801) are exact multiples of the desired frequency; thus the harmonics of 100 cycles would be 200, 300, 400, etc. cycles. Intermodulation products, which occur only when two or more frequencies are simultaneously reproduced, consist of various multiples of one frequency plus or minus multiples of another frequency; thus 100 and 1,000 cycles might combine to produce 1,100, 900, 2,100, 1,900, 2,200, 1,800, etc. cycles. Intermodulation distortion is generally much harsher to the ear than harmonic distortion.

Both of these forms of distortion are usually more irritating than a third common type, called transient distortion. The latter denotes the inability of a component to follow abrupt changes of sound. Thus, a speaker or power amplifier may continue for a brief but unpleasant while to reproduce a sound no longer being fed to it (Fig. 802).

The problems we shall be discussing in this chapter involve mainly the production of harmonic and intermodulation distortion.

Speaker problems

Particularly at low frequencies, the performance of the speaker in terms of distortion (as well as in terms of frequency response) depends on the enclosure in which it is mounted. This is one of

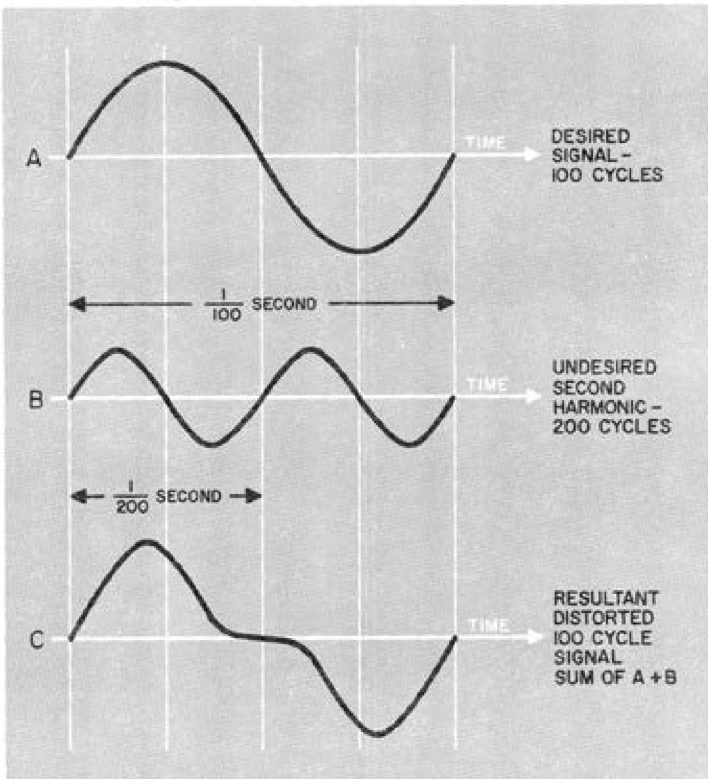


Fig. 801. *Example of harmonic distortion.*

the reasons for the popularity of the integrated speaker system, where enclosure and speaker are made for each other. If your preference is to purchase the speaker alone and install it in an enclosure of your own choosing and design, be sure to follow carefully the speaker manufacturer's recommendations concerning type of enclosure, dimensions, panel thickness and bracing, etc.

As already pointed out, intermodulation distortion is the worst type but can occur only when two or more frequencies are simultaneously reproduced. Thus a way of reducing intermodulation distortion is to have different speakers reproduce different frequencies. This (in addition to better frequency response) is the

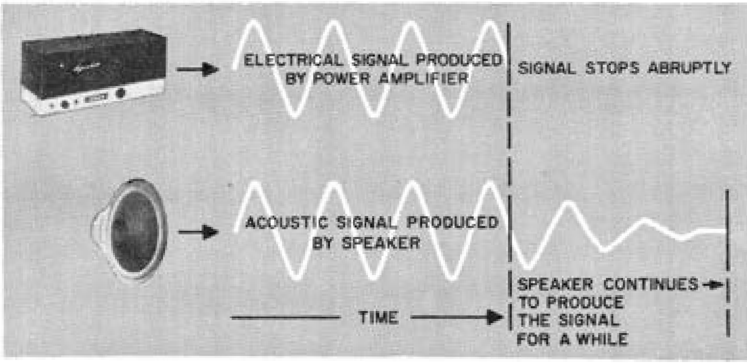


Fig. 802. *Example of transient distortion in a speaker.*

reason why most integrated speaker systems consist of at least two speakers within an enclosure. Typical is the two-way system comprising a tweeter for high notes and a woofer for the middle and low frequencies. There are three-way and even four-way systems available, further reducing the opportunity for intermodulation distortion.

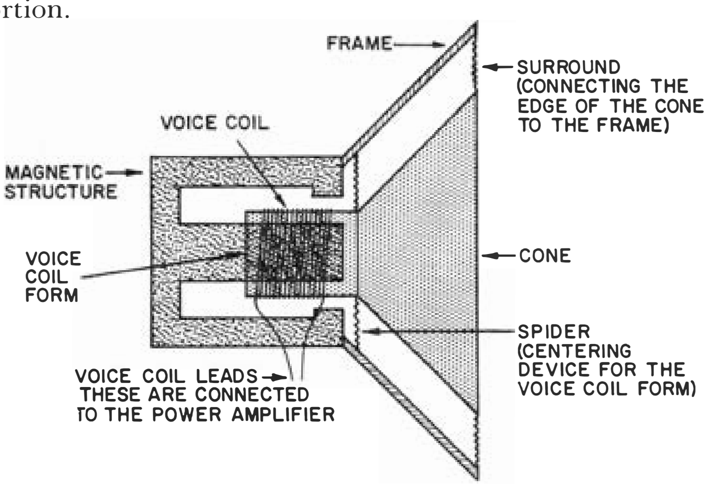


Fig. 803. *Basic structure of a speaker.*

As illustrated in Fig. 803, the electrical signal produced by the power amplifier is fed to the speaker's voice coil, which moves back and forth within a magnetic field. The voice coil in turn causes a cone or diaphragm to move, which engages the air and thereby produces sound. The air gap surrounding the coil is usually very small (for reasons of speaker efficiency), and, if the coil is not precisely centered, it may rub against the magnetic structure, producing a more or less grating distortion. The sound is often similar to the distortion produced by other components, so that a faulty speaker is not always immediately suspect.

When the trouble is identified, the usual course is to have the speaker repaired by an expert in this kind of thing. However, such is not always the case. An individual installed a fine wide-range speaker in the wall of his home, and it worked very well for a while. But that winter he complained of a slight distortion which would come and go. It turned out that the wall in which the speaker was mounted backed onto the garage, which was stone cold. With warmth in front and cold behind, especially on very frigid days, the speaker warped enough to cause slight rubbing of the voice coil. The evident cure was to mount the speaker elsewhere or provide heat behind.

Distortion may result from operating a speaker specifically intended as a woofer or tweeter outside its intended range, particularly if it is operated at high level. To illustrate, assume that a tweeter is designed to operate at frequencies above 1,000 cycles but that an attempt is made to have it reproduce down to 500 cycles. At 500 cycles the tweeter may be making such prodigious back-and-forth excursions as to produce a peculiar wail known as "cone cry." If a woofer is forced to operate above its intended range, the result may be a fuzziness due to what is called "cone breakup." Hence, if you choose to put together a two-way or three-way system instead of buying an integrated system all ready to go, give careful consideration to your choice of speakers and their respective operating ranges.

A faulty lead between the power amplifier and speaker may produce an effect akin to distortion. Due to excessive twisting or bending, a lead may have broken; but the broken ends may be barely touching, so that the sound continues in a ragged fashion. Here you can put your continuity checker to work, as illustrated in Fig. 804. If the bulb flickers, this indicates an intermittent connection; jiggle the lead or pull on it a bit as you make the test. It may be necessary to replace all of the lead or just a section.

If you have made a sloppy connection either to the speaker terminals or to the power amplifier's output terminals, so that a stray strand connected to one terminal is brushing another terminal, there is a partial short-circuit which may cause the sound to appear distorted.

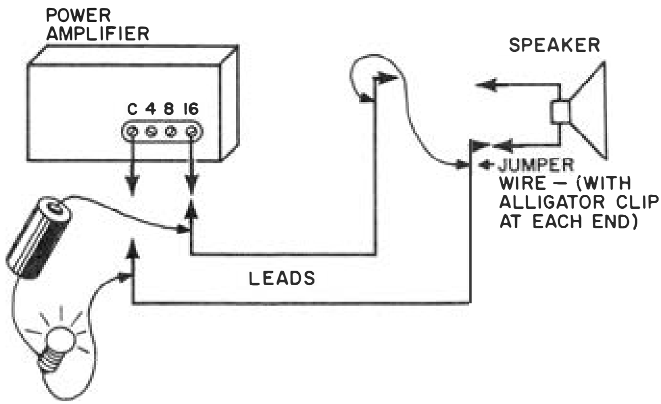


Fig. 804. *Checking for a break in the leads between the power amplifier and the speaker. Refer, also, to Fig. 901, p. 116 and Fig. 908, p. 123.*

Power amplifier problems

If you like to operate your speaker at high levels, the power amplifier may be forced into distortion for one or more of the following reasons

1. If the amplifier has a relatively low power rating (under 15 watts or so) and the speaker is relatively inefficient (some speakers require 10 times as much power as others to produce the same sound level) , the amplifier may simply be unable to supply the required power at low distortion.
2. The amplifier may have ample power through most of the audio range, but it may be inadequate at the bass end. For example, a moderate-priced amplifier may be rated at 30 watts, and quite likely it will produce 30 watts at mid-frequencies. But its power capability may shrink at frequencies below 100 cycles. Possibly it can produce only 5 or 10 clean watts at the low end. If your preference is, say, for loud organ music, you may have trouble with such an amplifier.
3. The speaker and amplifier may be mismatched. Ordinarily, the rated impedance of the speaker should be matched to the output impedance of the amplifier. Thus if the

speaker is rated at 8 ohms, it should be connected to the 8-ohm tap of the amplifier (Fig. 805). Matching permits maximum power transfer at minimum distortion from the amplifier to the speaker. Occasionally you may find that connecting the speaker to a tap higher in impedance than

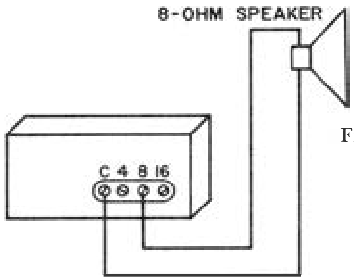


Fig. 805. *Matching the speaker and power amplifier.*

its own rating gives improved results at low frequencies—for example, connecting an 8-ohm speaker to the 16-ohm amplifier terminal. The reason is that speaker impedance typically rises at low frequencies (Fig. 806). Hence, at low frequencies a speaker nominally rated at 8 ohms might actually have an impedance closer to 16.

Distortion may of course be due to tube troubles. The prime suspects are probably the output tubes. If you check by substitution, it is preferable to use matched tubes. In some power amplifiers, when installing new tubes, it may be necessary to adjust one or more controls: bias (governing the total amount of current drawn by the two tubes), bias balance (causing each output tube to draw an equal amount of current) and dynamic balance (causing equal audio signals to be fed to each tube). In fact, whether or not you're putting in new tubes, it is a good idea to check these adjustments every few months if you want minimum distortion and maximum tube life. A number of power amplifiers include a meter so that you can make these adjustments yourself. On the other hand, many amplifiers are designed so they amplify pretty well without these adjustments.

A weak rectifier may be responsible for distortion. Generally the rectifier (which supplies high-voltage direct current) is a tube, and in this case is simple and inexpensive to check and replace. But some amplifiers use a silicon diode instead of a tube. Since silicons are often neither simple nor cheap to check and replace, you will probably need a technician—unless the diodes are mounted where they are easily accessible and plug in and out of the amplifier. Fortunately, silicon diodes are quite durable and require replacement less often than tubes.

A faulty capacitor or resistor—one that has shorted, opened, or changed in value is a common source of distortion in electronic components. The repair is up to the technician, but the prevention of this kind of trouble is partly up to you. Overheating is one of the principal factors that cause resistors and capacitors to go

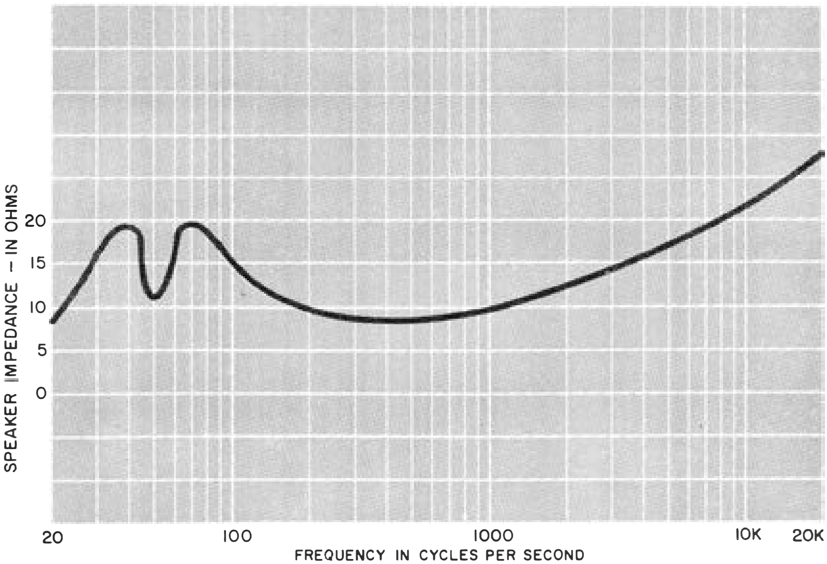


Fig. 806. *Typical variation of speaker impedance with frequency (8-ohm speaker in a bass-reflex enclosure).*

bad. To the extent that you provide good ventilation for your amplifier you reduce the chance of a resistor or capacitor becoming defective. The voltages to which resistors and capacitors are subjected also affect their life. A high-quality amplifier provides a substantial safety factor by using resistors and capacitors with voltage ratings substantially over the values they will actually encounter in the circuit. On the other hand, if your line voltage is too much above the nominal 117 volts, this will increase the operating voltages in your amplifier and lower the safety factor. If you have reason to suspect that your line voltage is high (perhaps light bulbs are blowing out with greater frequency than normal) call your electric company to run a check.

A well designed power amplifier is quite tolerant as to the type of speaker you connect to it, but a mediocre amplifier may not be. If connected to an electrostatic speaker, it may go into oscillation, producing a supersonic frequency which is inaudible in itself but may well cause appreciable distortion due to interaction with audible frequencies. The same thing might happen if you were to use a very long run of cable from a mediocre amplifier to a conventional speaker; the capacitance between the two leads of the speaker cable might "upset" the amplifier.

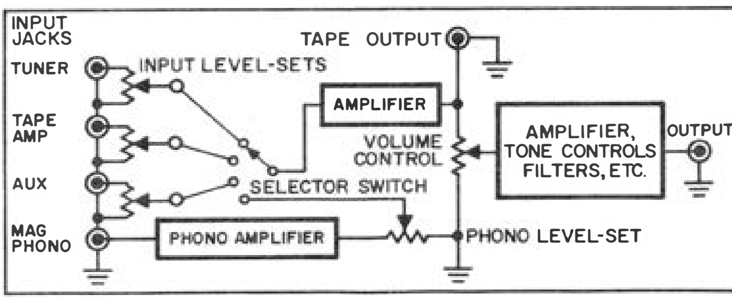


Fig. 807. *How input level sets are incorporated in a typical preamplifier.*

Preamplifier problems

When a preamp has to contend with a large incoming signal, appreciable distortion may occur under certain conditions. Suppose that a preamp can convert an input signal of 0.2 volt (say from a tuner) into 2 volts output having less than 1% distortion. If more than 0.2 volt is fed in, the output voltage will rise proportionately. But distortion too will rise, perhaps disproportionately. In some preamps, distortion may rise to 4% or 5% by the time the output signal reaches 3 volts, which is unsatisfactory for high fidelity.

Thus there may well be a problem of reducing the input signal. If you get more than enough volume at quite low settings of your volume control (below 11 o'clock or so), this suggests that too much signal is coming into your preamp. A tuner, tape recorder or TV usually has a gain control that can be used for signal reduction. Or the preamp may have an input level set (Fig. 807). Either approach is excellent.

Another means of reducing the input signal is the preamp's volume control, but here is a possible danger. Unless this control precedes all tube stages in the preamp, there is risk of considerable distortion (Fig. 808). While the volume control can prevent the *following* tube stages from being overloaded by a large input signal,

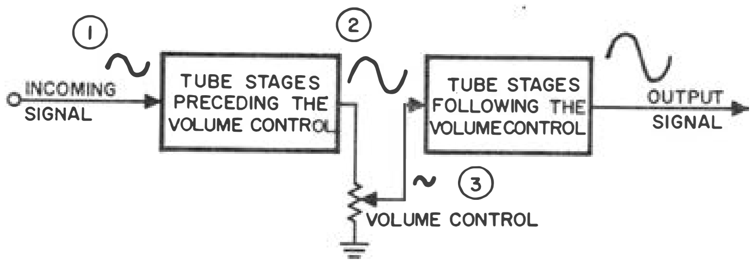


Fig. 808. *How a volume control can be ineffective in controlling distortion in the early tube stages due to an excessive input signal.*

it cannot prevent the *preceding* tube stages (if any) from being overloaded. That is why preamps which locate the volume control after one or more tube stages usually have (or should have) input level sets for reducing the input signal and preventing distortion.

You can also have problems with the preamp's output signal level. Many or most power amplifiers have an input level set which, as illustrated in Fig. 809, can reduce the signal produced by the

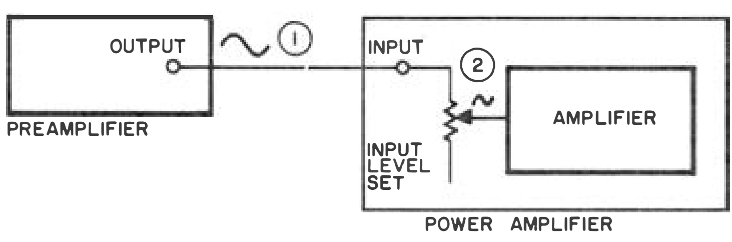


Fig. 809. *Role of the input level set in the power amplifier.*

preamp. (The purpose is to prevent the power amplifier and in turn the speaker from being driven too hard, with possible damage to either one; also, to reduce noise produced by the preamplifier). The farther down you turn the power amplifier's level set, the harder the preamp has to work to supply enough signal to drive the power amplifier and in turn the speaker. Thus a preamp may be forced to work so hard that it is turning out much more than 2 volts, at considerable distortion.

Since there are limits as to the amount of signal which the preamp can accept and deliver with low-distortion, you may have to experiment a bit to find an optimum combination of the various controls that affect signal level. Fortunately, a good combination of settings is seldom critical, and usually covers a tolerable range. With some preamps this range of satisfactory settings is greater than with others. Although many preamps can turn out a low-distortion signal only up to 2 volts, others can do so up to 5 volts, and a few even up to 10 volts.

Be wary about operating your preamp with both the volume and bass controls turned up high. A high setting of the bass control may mean that your preamp is trying to produce 5 to 10 times as much signal at 50 cycles as at 1,000. Although the preamp may be causing very little distortion at 1,000 cycles, it may be forced to generate a good deal of distortion at 50 cycles due to the bass boost. The same kind of thing can happen in the treble range if the treble control is turned way up at the same time as the volume control.

Tuner problems

For minimum distortion it is vital that an FM tuner be aligned with extreme care, either at the factory or subsequently by a *very* competent technician. A few of the top-flight audio stores, which compete on the basis of quality and service rather than price, make it a practice to check carefully the alignment of every FM tuner they sell.

Good alignment becomes even more important if you try to bring in signals from stations at substantial distances. And if the signal is a multiplex (stereo) one, the importance is further emphasized. Except in the case of a few tuners built from kits and designed for alignment by the builder, FM alignment is far beyond the scope of the average audiophile and should be entrusted only to an expert technician or to the tuner manufacturer.

For clean reception of distant signals, a high-quality antenna specifically designed for FM has much to contribute. An antenna rotor can make a further contribution if the distant stations are at various points of the compass. Another device that can play a useful role in achieving low distortion is an antenna amplifier to build up the signal before it reaches the tuner. This is an electronic device mounted either on the antenna itself or a few feet away (just inside the house).

Tuner drift is a frequent source of distortion. As the FM tuner warms up, certain critical parts tend to change slightly in value, causing partial detuning and consequent distortion, particularly on loud sounds. Be careful not only to tune as precisely as possible to a given station, but recheck your tuning once or twice as the tuner warms up. If the tuner continues to drift for more than 15 minutes, a visit to the audio technician is recommended.

To minimize the effect of inaccurate tuning and drift, many tuners incorporate afc (automatic frequency control) to lock onto a desired station. They also usually enable you to "defeat" the afc so that you can "capture" a weak station which is right next to a strong station on the dial. Once you have captured the weak station, it is generally desirable to restore the afc to minimize distortion.

Distortion is not always the tuner's or operator's fault. Fairly often it is the fault of the broadcast station. The process of impressing an audio signal on a radio wave is called modulation. The amount of modulation is limited by law, and FM tuners are generally designed to produce low distortion so long as modulation is within the limits set by law. Nevertheless, FM stations sometimes overmodulate, with resultant distortion in the tuner. A few tuners are designed to cope with such overmodulation. But most aren't. What you might do, if you suspect overmodulation by the station—which imparts a rasping quality on loud sounds—is to compare notes with several friends. If you all share the same experience with a variety of tuners, you can take your complaint first to the offending station and then to the Federal Communications Commission.

Distortion may also be the station's fault because of a poor broadcast signal, owing to such things as faulty equipment, worn recordings and carelessness.

Phono problems

Excessive signal produced by the phono cartridge may cause distortion. Whereas the average magnetic pickup delivers 10 to 20 millivolts at most, some pickups turn out a good deal more, perhaps enough to overload the phono amplifier and "dirty" the sound. Therefore some preamps provide separate input jacks for high-output cartridges and for low output ones; these jacks are respectively marked "high" and "low." If you feed a high output cartridge into the "low" jack, you risk distortion.

If your preamp has but one input for magnetic cartridges and your cartridge appears to be overloading the preamp, a simple modification can be made to reduce the cartridge signal (Fig. 810). The load resistor across the phono jack is replaced by two resistors in a series, with a total value equal to that of the original resistor. The two resistors form a voltage divider, and the circuit lead that

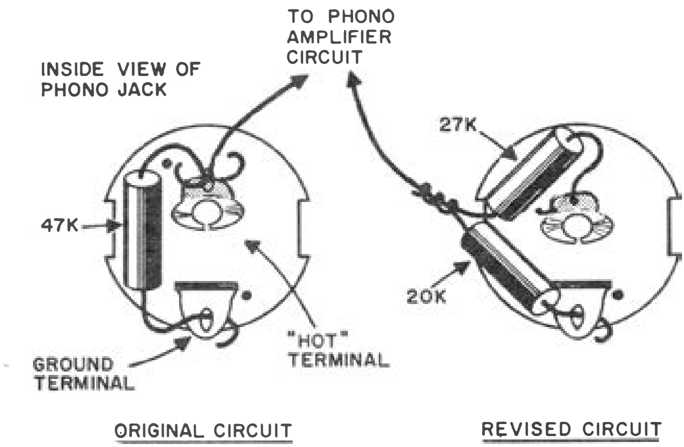


Fig. 810. *Reducing the signal from a magnetic phono cartridge.*

formerly went to the hot terminal of the input jack now goes to the junction of the two resistors. The decreased voltage is proportional to the value of the lower resistor (the one that goes to ground) divided by the total resistance. Thus in Fig. 810 the reduced voltage is 20/47 of the original voltage, that is, somewhat less than half. Other resistor combinations can be used to decrease the input voltage more or less.

A ceramic pickup can also overload the phono amplifier. If this pickup is fed via an adapter into the magnetic phono input, the cure is the same as described in connection with Fig. 810.

When the ceramic pickup is fed into a high-level input, again the method of Fig. 810 can be used. Otherwise, you can wire a capacitor of suitable value across the leads to the cartridge. At the same time this capacitor will augment the cartridge's bass response, as was discussed in Chapter 6. Quite possibly you can kill two birds with one stone—achieve both a needed increase in bass and a needed reduction in signal level—by putting a capacitor across the cartridge. As stated in Chapter 6, a capacitor between .002 and .003 microfarad is apt to be suitable. The larger the value of the capacitor, the greater is the signal attenuation; and the greater is the increase in bass response (up to a point).

What is called the lateral angle of the cartridge stylus may be responsible for distortion (and also for poor separation between channels when playing a stereo record). With the cartridge in playing position and viewed head-on, the stylus should be perpendicular to the record (Fig. 811). You can check this by placing a

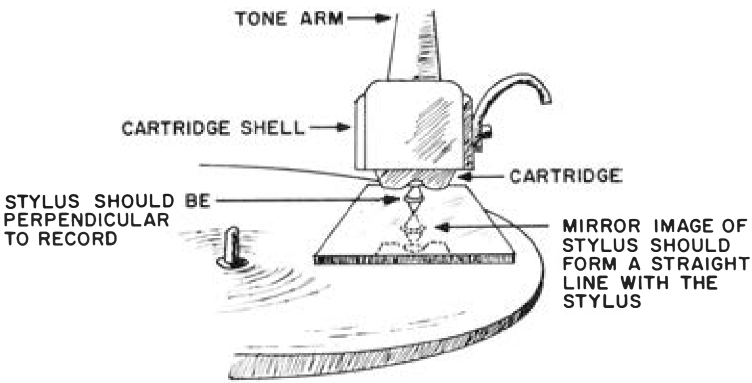


Fig. 811. *Checking the lateral angle of the cartridge stylus. Size of stylus has been deliberately exaggerated.*

mirror face up between the phono disc and the stylus. If the stylus and its mirror image form a straight line, the lateral angle is correct. If they don't, check whether the stylus appears bent or otherwise dislocated with respect to the cartridge. If it is, the stylus probably should be replaced. If not, the angle of the entire cartridge should be adjusted within the tone arm, by means of a shim under one of the mounting screws. Some tone arms provide means for slightly adjusting the tilt of the arm and thereby the stylus angle.

Ideally, the axis of the cartridge and stylus should form exactly a 90° angle with the radius of the record (Fig. 812). The reason is that, when a record is made, the cutting stylus forms such an angle. If the playback stylus forms a different angle, the deviation is called tracking error, and the result is distortion. In practice, it is impossible for the playback cartridge to maintain zero tracking error throughout the playing of a record. However, tracking error can be reduced to virtual insignificance by careful design of the shape of the tone arm, precise location of the arm with respect to the turntable center and precise mounting of the cartridge in the arm for correct overhang. Overhang (Fig. 813) is the distance from the stylus tip to the turntable center when the cartridge is directly over the center.

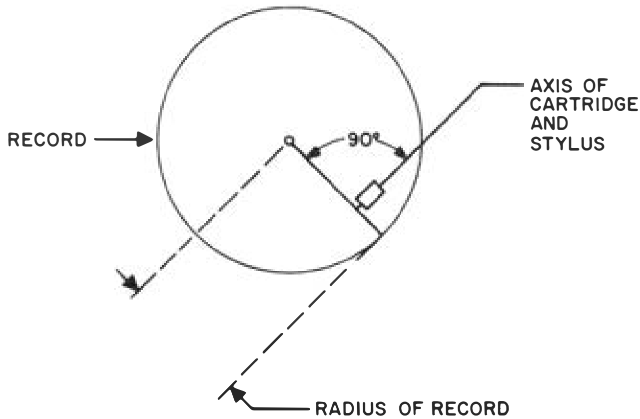


Fig. 812. *Ideal orientation of the cartridge axis with respect to the record.*

If you yourself install the tone arm on the turntable, make sure that everything is mounted strictly in accordance with the manufacturer's instructions. If the arm and cartridge are installed by

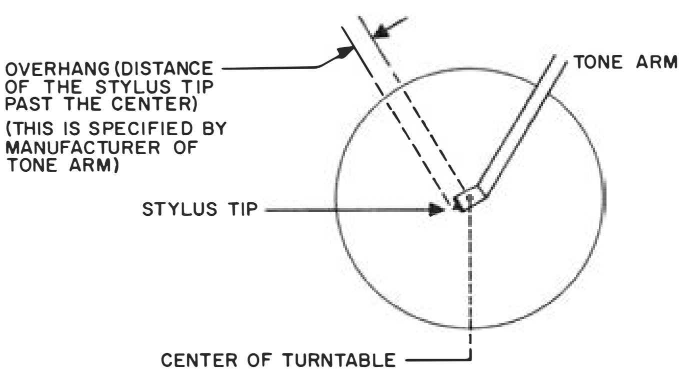


Fig. 813. *Stylus overhang.*

your audio dealer, it won't hurt to check his work. All you need is an accurate ruler and the instructions that come with the arm.

The stylus can follow the record groove only if the tone arm exerts some downward pressure or tracking force. This presents a dilemma. Too little tracking force will allow the stylus to bounce around in the groove, perhaps even out of the groove, causing distortion, especially on loud passages. Too much tracking force causes undue record wear. Between these extremes is an optimum tracking force, which varies with the individual cartridge and the

Fig. 814. *Acoustic Research beam balance for measuring tracking force.*

tone arm used. Obey the cartridge manufacturer's recommendations as to proper tracking force.

Every tone arm worthy of the term high fidelity, whether on a changer or manual turntable, has a tracking-force adjustment. The force can be measured by various devices on the market, such as the beam balance illustrated in Fig. 814. Some tone arms have calibrated markings to indicate tracking force as you move a weight or tighten a spring. However, these markings are not always very accurate, and you are apt to be better off with a measuring device such as that in Fig. 814.

The cartridge manufacturer usually states the recommended tracking force as a range rather than as a precise figure; for example, he may recommend 1½ to 2½ grams. You may find least distortion when the tracking force is at or toward the top of this range. You may even find that, on some records, it is necessary to exceed the recommended limit to get away from discernible distortion (called *tracing*—not tracking—distortion). Don't be afraid to exceed this limit if your ear tells you to do so. Low distortion is more important than record wear; few phono records are worn out in the home from use. In fact, to the extent that the stylus bounces around in the groove because of insufficient tracking force, it is apt to cause more record wear than if it were faithfully tracing the groove at somewhat elevated force.

In terms of the radius of its tip, the cartridge stylus generally comes in one of four sizes: 3-mil (thousandths of an inch) for playing 78-rpm records, 1-mil for mono LP records, and 0.7-mil or 0.5-mil for stereo discs. If a cartridge with a 3-mil stylus is inadvertently used to play an LP disc (mono or stereo), it cannot seat itself sufficiently deep in the narrow groove to follow the groove accurately, and distortion results. Conversely, if a small-radius stylus is used on a 78-rpm record, the groove is so wide that the stylus hits bottom and rattles about, causing distortion. The same kind of thing often (though not always) happens when a 0.7-mil or 0.5-mil stylus is used on a mono disc, which has a somewhat wider groove than do stereo discs.

For stereo, the ideal stylus radius is 0.5 mil. But this fine a tip tends to accentuate record wear unless tracking force is very light, which in turn is feasible only with a very skillfully designed and balanced tone arm. As a compromise, therefore, particularly for use with record changer arms, most stereo cartridges are equipped with a 0.7-mil stylus. In some instances the compromise radius is 0.6 mil. But if you have the finest of turntable equipment and wish the cleanest reproduction, use a cartridge with a 0.5-mil stylus.

Distortion is caused by dirt in the record groove and dirt on the stylus tip. Because of the light tracking force used nowadays, often below 2 grams, even on record changers, it doesn't take much to accumulate enough fuzz on the stylus to lift it from the groove and cause painful distortion. Therefore clean and destaticize your records with suitable agents, and clean the stylus with a moderately stiff brush after every playing of one side of a record. Of course, if you are using a changer, you can't clean the stylus as often; on the other hand, you can pile up fewer records to shorten the intervals between cleaning.

Wow and flutter may be considered forms of distortion. Wow consists of slow variations in speed. Flutter consists of rapid variations which are not discernible in themselves but add a fuzzy or grainy quality to the sound. Every turntable has some wow and flutter. To help keep these satisfactorily low, follow the manufacturer's instructions with respect to periodic cleaning, lubrication and checkup.

Like wow and flutter, rumble is an undesirable ingredient added to the sound. It is the result of electrical and mechanical imperfections in the turntable. To keep rumble at a minimum, follow the cleaning, lubrication and other maintenance measures recommended by the manufacturer.

Tape recorder problems

In seeking that difficult thing to attain in a tape recorder, a high signal-to-noise ratio, the operator is tempted to record at excessively high level, which breeds distortion. Just a slight increase beyond maximum permissible recording level, as indicated by the record-level indicator, can produce a relatively great increase in distortion. Keep a careful eye on that indicator.

Of course, it is possible that the indicator is miscalibrated and doesn't warn you in time that you are recording at too high a level. If you heed the indicator but nevertheless get too much distortion, a visit to the technician may be in order. On the other hand, if the indicator is a VU (volume unit) meter, the fault may be yours. Keep in mind that a VU meter reads average rather than peak level. Usually, VU meters are calibrated with a safety factor of about 6 db (4:1 ratio in terms of acoustic power) to allow for the difference between the meter reading and the actual peak level. But, as illustrated in Fig. 815, the peaks of the audio signal may be as much as 20 db (100:1 power ratio) greater than the average level on certain kinds of program material. Hence you must learn

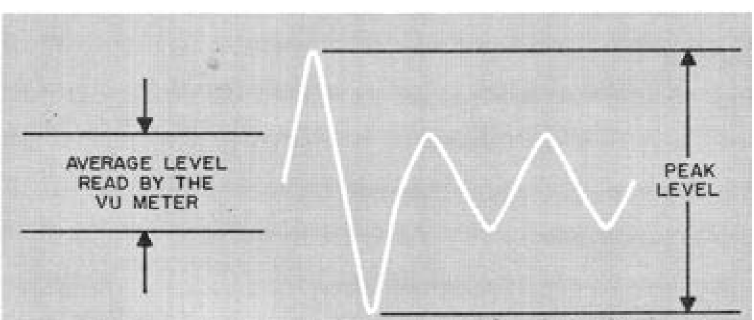


Fig. 815. *Comparison of peak and average level of a typical audio signal.*

through experience to interpret the VU reading, taking into account the kind of material being recorded. Because of this problem of interpretation, some home recordists prefer a magic-eye indicator, which follows peak signals more accurately.

To keep tape distortion far lower than it would otherwise be, a high-frequency "bias" current, usually between 50,000 and 100,000 cycles, is fed to the record head along with the audio signal. Decreasing this current below the optimum value specified by the tape recorder manufacturer can cause a sharp rise in distortion (Fig. 816). At the same time, treble response increases. If there are simultaneous symptoms of an increase in distortion and in treble response, it is logical to suspect insufficient bias current. Try replacing the oscillator tube. If this doesn't help, your problem requires a technician.

Excessive friction between the tape and the heads may cause the tape to vibrate like a violin string rubbed by a bow. This vibration causes distortion. Such friction may be due to several things: (1) dirt or tape oxide on the heads, (2) a poor-quality tape containing insufficient lubricating material, (3) a tape that has lost its lubricant or moisture through use or storage conditions, (4) too much force exerted by the pressure pad against the heads, (5) sticky pressure pads that require cleaning or perhaps replacement.

Cleaning the heads at fairly frequent intervals (about every 8 hours) with alcohol or a specific preparation, and use of high quality tape, are obviously called for. Several agents are on the market for lubricating tape. Moisture can be restored to a dried-out tape by keeping it in a container along with a moist sponge for about 24 hours. Special lubricants are available for application to the heads and tape guides. *These should never be applied to the capstan and pressure roller.*

Adjustment of pressure-pad force will probably require the services of a technician. Replacement of pads, however, is simple enough. With a fine pencil draw an outline on the pressure-pad mount to show where the pad is presently located. Strip off the pad and replace with pad material, cut to size and shape. The material can be had from an audio supply house, from the tape recorder manufacturer or from one of his authorized service agencies.

Depending on the formulation of the magnetic oxide, tapes differ with respect to distortion. Among top-quality tapes made by firms of established reputation, these differences tend to be minor. However, the difference might be appreciable between a high-quality tape and a cheap brand.

As with the phono turntable, you can play your part in keeping wow and flutter to a minimum by following maintenance instructions. In particular, clean the capstan and pressure roller period-

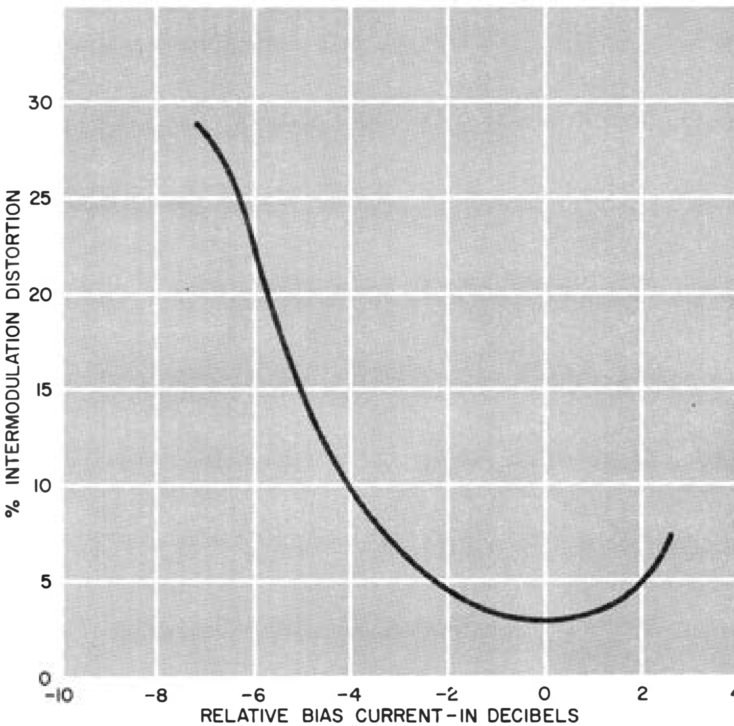


Fig. 816. *Variation of intermodulation distortion with changes in bias current.*

ically, using alcohol or other specific agent. However, a tape transport is much more complex than a phono turntable, and many more things can go wrong and thereby increase wow and flutter. Various mechanical adjustments, involving spring tension and the like, are apt to be quite critical and best left to an experienced technician.

Most home tape recorders are self-contained in the sense that they include a small power amplifier and a speaker. At the same time, they generally provide for feeding an external audio system and have one or two output jacks. If there are two jacks, these are typically marked "external speaker" and "external amplifier." To feed your audio system, take the signal from the output jack marked "external amplifier," because the signal will contain least distortion at this point. The reason (Fig. 817) is that the signal at this jack is taken prior to the tape machine's output stage (output tube and transformer), which is usually not designed to meet high fidelity standards and introduces appreciable distortion (as well as loss of low and high frequencies). If the tape machine has but one

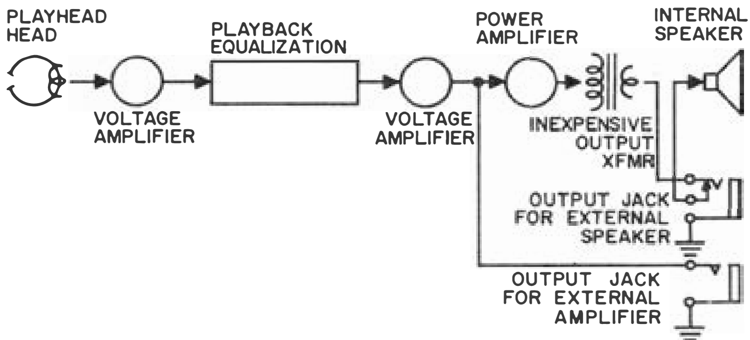


Fig. 817. *Method typically used in home tape recorders for supplying the playback signal to an external speaker or an external amplifier.*

output jack, determine whether the signal is taken before or after the output stage. If taken after, it may be feasible for a technician to install an extra output jack so that the signal is derived from a point prior to the output tube.

Chapter 9

No Sound (or Weak Sound)

UNTIL NOW WE HAVE BEEN CONCERNED WITH IMPERFECTIONS IN THE sound—with hum, noise, poor bass response, poor treble and distortion. In this chapter we are concerned with the absence of sound altogether, either initially when an audio system is installed or after it has been in satisfactory operation for a time. As always, the first step is to ascertain which component of the system is responsible. By referring back to Chapter 3, on the art of substitution, you can by a logical process of elimination quickly ferret out the responsible component.

Lack of alternating current

High-fidelity components are generally dependent on house current for operation. The principal exceptions are the speaker and transistorized components operated from a battery, such as a preamplifier for a microphone or low-output phono cartridge. And the speaker is not a total exception, because electrostatic speakers are dependent on house current. By and large, then, the lack of ac is a prime reason for sound failure and the first thing to be investigated.

The surest way to ascertain whether a component is receiving current is to see if the tubes light. If the component has no tubes and is transistorized, it still probably has a tattle-tale pilot lamp. On the other hand, the fact that the pilot lamp fails to glow is not a sure indication of ac failure—just the lamp may have gone out. Replacement of the pilot lamp will then tell the story.

Replacing a pilot lamp requires removal of the component from its cabinet or other resting place; also removal of the component's top or bottom cover. This may be necessary also to see whether the tubes light. For this and other reasons, the initial step you may want to take is to find out whether ac is *available* to the component in question. First check the house socket that provides ac to your entire system. This is easily done by plugging a lamp, radio or other appliance into the socket. Next, check the socket that furnishes ac to the component in question, if this is other than the house socket. For example, the power amplifier is usually plugged into an ac receptacle contained in the preamplifier, or the tuner or tape recorder may be so connected.

If the house socket is "dead," check the house fuse. If replacement of the house fuse doesn't help, call an electrician or find a house socket that is in working condition. As a safety measure, call the electrician anyway. If the dead socket proves to be in the preamplifier, try another socket if available; preamplifiers generally provide several ac outlets. If all are dead, your problem belongs to an audio technician, unless you built the preamplifier from a kit. In that case, check your work carefully.

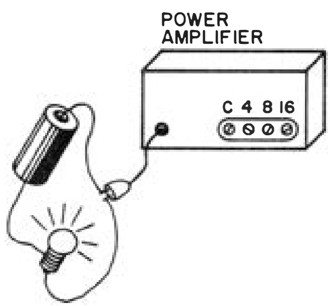


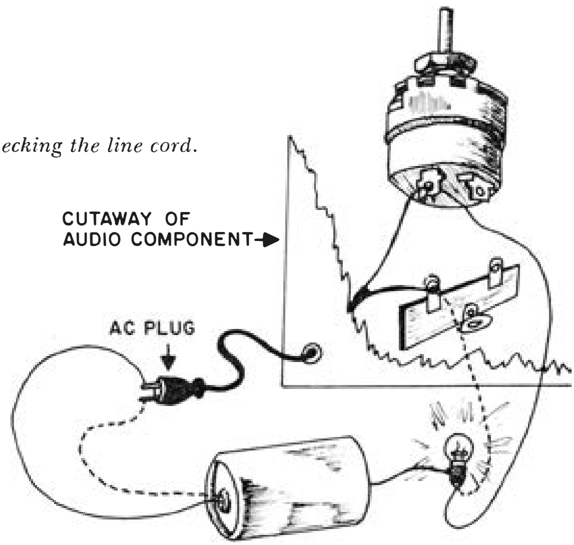
Fig. 901. *Checking 117-volt ac continuity in a power amplifier.*

Let us assume that all the sockets you have tested prove to be in working condition; that is, ac is *available*. The next thing is to check whether the component in question is capable of *drawing* current. Assume that this component is the power amplifier. As shown in Fig. 901, use your continuity checker to ascertain whether there is a continuous electrical path between the two prongs of the component's line plug. This path is from one prong, through the component and out the other prong. The component's on–off switch should be in the on position when you do this checking.

If the bulb of your continuity checker lights, this tells you there is a continuous path for the ac, and you have to look elsewhere than the current supply for the reason for sound failure. But if there is no continuity—the bulb doesn't light—check:

1. *The component's fuse,* if there is one. It may have burned out. Fuse failure is usually obvious to the eye but, if there is any doubt, test the fuse with your continuity checker. (As will be discussed in the section on fuse replacement, don't hastily test the old fuse by putting in a new one.)
2. *The component's line cord.* It may be faulty. Checking the cord requires going inside the component, but the task is an easy one. Each lead of the line cord terminates at a lug that you can find by eye on a terminal strip, switch, fuse, etc. As shown in Fig. 902, test the line cord by placing the continuity checker between one prong of the line plug and

Fig. 902. *Checking the line cord.*



each of the terminating lugs; then between the other prong and these lugs. If the bulb fails to light while the continuity checker is connected to one of the prongs, the line cord has a break in it and must be replaced.
3. *The prongs of the line plug.* They may not be making contact within the socket into which the plug is inserted. File these prongs clean and bright. Twist them slightly in opposite directions, as illustrated in Fig. 903, to insure a secure fit in the socket.
4. *The ac switch of the component in question.* It may be defective. Fig. 904 shows the typical path for alternating

current through an audio component—through a switch, fuse and power transformer winding. Make sure that the on–off switch is in the *on* position. Place the continuity checker across the switch terminals, which you can locate by eye by tracing the line cord. Failure of the bulb to light indicates a faulty switch.

Blown fuse

A blown fuse is one of the first things to be investigated. If visual inspection or your continuity checker shows that it has blown out, don't immediately put in a new fuse and turn the equipment on.[1] Chances are that you will simply blow one fuse after another. And

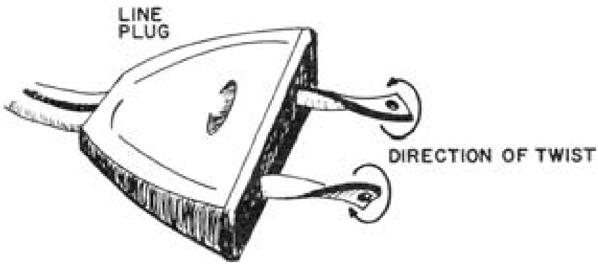


Fig. 903. *Twist the prongs of the line plug to secure a tighter fit in the socket. Amount of twist shown here is greatly exaggerated. Slight twist is generally all that is required.*

don't try putting in a fuse with a much higher amperage rating, for example a 5-amp fuse in place of a 2½. If you do, you risk damaging the component.

Before replacing the fuse, remove all the tubes. If possible, have them checked in a tube tester, because there is a good chance that a shorted tube is drawing excessive current and blowing the fuse. If the tubes all check good, or if you are going to install one or more new tubes, don't put them back yet. Replace the fuse and turn the equipment on. If the fuse blows again, this may likely be due to a faulty power transformer, and your problem should be referred to an audio technician. If the fuse holds up, replace the rectifier tube first. If the fuse blows again, this indicates a short circuit caused by something other than a tube, for example a faulty capacitor or resistor. If the metal plates of the rectifier tube glow red before the fuse gives way, this corroborates the indication of a short circuit. And you require the services of a technician.

[1] Unless you know that the fuse blew due to a voltage surge, as might happen during an electrical storm.

If the fuse holds, replace the other tubes one at a time. When all the tubes are replaced, the fuse may give way again. Check whether the new fuse was an *exact* replacement both as to *type* and *value*. A slow-blow type of fuse is often specified, because this can withstand the momentary current surge that occurs when a component is first turned on.

In the case of a power amplifier blowing fuses after a short period of operation, note whether the metal plates of the output tubes were glowing red. If they were, insufficient negative bias voltage was applied to these tubes. Check your instruction manual on how to adjust the bias; a number of power amplifiers incorporate a meter so that the layman can make the adjustment correctly. If there is no meter, you can use a stop-gap procedure at the risk of somewhat greater distortion than if bias were adjusted precisely: Adjust the bias control to the point where the output tubes just stop glowing red.

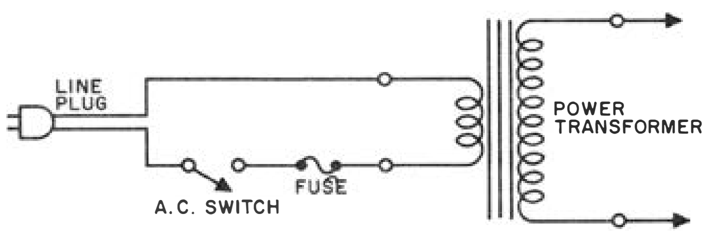


Fig. 904. *Typical 117-volt ac path to the power transformer in an audio component.*

Occasionally it may turn out that, even though all the tubes are good and there is no indication of a short circuit or anything else to make the component draw excessive current, the fuse continues to blow—not necessarily immediately but at frequent intervals. Here the fault may be excessive line voltage.

The voltage at your house socket should normally be about 117, and audio equipment is designed around this value. However, on occasion, the line voltage may go as high as 125 and possibly 130 volts, depending on such factors as the distance between your house and the generating station, the distance to the outdoor transformer supplying current to your house, the time of day or night, and possible lack of care on the part of the generating station. When subjected to appreciably more than 117 volts, your equipment may blow fuses. If there is reason to suspect excess line voltage, ask your electric company to check this out. Perhaps you can borrow an accurate ac voltmeter to check it yourself.

If the line voltage is only slightly high but this is nevertheless causing fuse failure, you might try using a fuse with slightly higher amperage rating than the original one, but no more than 20% higher. Thus you might use a 3½-amp fuse in place of a 3-amp. Before changing to a higher value, note whether the original fuse was a slow-blow type. If not, first see whether a slow-blow fuse will solve your problem.

Faulty tubes

A weak or dead tube can be responsible for partial or total sound failure. The most likely suspect is the rectifier tube, which is required to handle more current than any other tube and consequently is usually the first one to go bad.

In a stereo component there is usually complete duplication of all tubes except the rectifier. Hence if both channels give trouble, the rectifier tube is the one to suspect. But if only one channel is weak or out, tubes other than the rectifier are suspect. In the case of the power amplifier, the prime objects of suspicion then are the output tubes, which like the rectifier tube are very hard workers.

A dead tube doesn't always mean total absence of sound. It is sometimes possible for the signal to pass around a dead tube, resulting in a sharply reduced but audible signal.

The fact that a tube is lit doesn't signify that it is in good working condition. This merely shows that the tube's heater is still working. On the other hand, so far as audio tubes are concerned, failure to light is a sure sign that the tube is bad. (Be careful here: Viewed in a bright light, a tube may seem not to be lit.)

Tube life depends a good deal on the amount of ventilation received. When mounting components in a cabinet, bookshelf, closet or other space, provide as much air space around each component as possible. There should be a minimum of 4 inches of air above each component. Stacking one component on top of another, particularly if the lower one is the power amplifier, is bad practice. If the components are rather tightly confined in a cabinet or other limited space without free air flow, ventilation can be improved by installing a small fan, which can be plugged into one of the preamp's switched ac sockets so that the fan goes on when the rest of the equipment is turned on. Conventional fans make enough noise to interfere with listening pleasure. Buy a quiet fan, especially made for the purpose at hand, at an audio store. Or you can quiet a conventional fan by reducing its revolutions per minute to half or a quarter of normal speed. This is done (Fig. 905), by

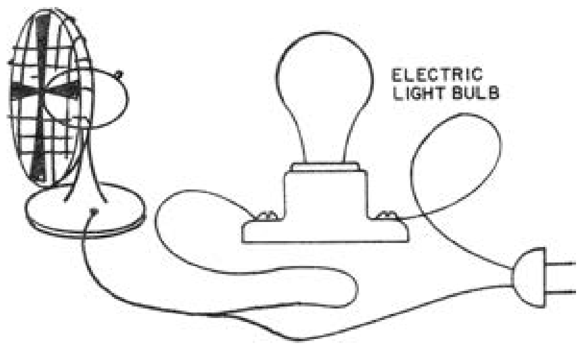


Fig. 905. *Using a small light bulb to reduce speed and noise of a fan.*

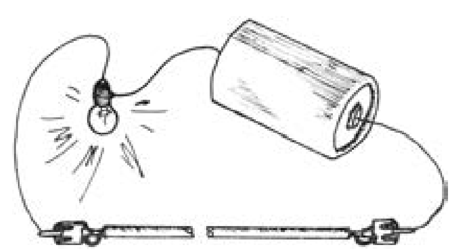
putting the fan in series with a small light bulb having a rating between 15 and 40 watts.

Most audio components are designed to be mounted horizontally, with the tubes standing vertically. Since heat rises, the tubes are therefore in the best position for heat to escape, thereby maximizing tube life. If you mount a component vertically rather than in the normal manner, this is likely to inhibit the escape of heat and shorten tube life.

Cables

Open or shorted cables can account for sound failure. In the case of shielded cable, a break in the ground (shield) lead will usually make itself well known by producing a pronounced hum. Perhaps the easiest way to check for a faulty shielded cable is by

Fig. 906. *Testing the "hot" lead of an audio cable for continuity.*



substituting a fresh cable between components, for example between a tuner and preamplifier. Lacking a fresh cable, you can put your continuity checker to work (Fig. 906). Test the hot (inner) lead of a shielded cable by connecting the continuity checker to each end of this lead. If the bulb lights, there is a continuous electrical path. To test for a short circuit between the hot and ground leads, use the continuity checker as in Fig. 907. Connect one side of

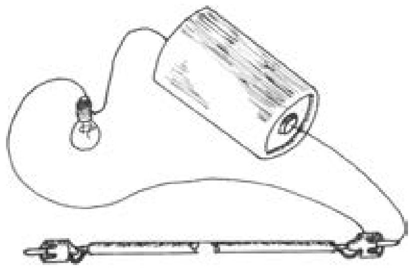


Fig. 907. *Testing an audio cable for a short circuit between the "hot" and ground leads.*

the checker to the hot lead and the other side to the shield. The bulb will light in case of a short.

Fig. 908 illustrates the procedure for checking the leads to the speaker. Disconnect these leads from both the amplifier and speaker. At one end, place a connecting wire between the two leads. Place the continuity checker across the other end of these leads. If the bulb lights, there is no break. To test for a short circuit between the speaker leads, remove the connecting wire and connect the continuity checker to these leads. The bulb will light if there is a short.

Speaker problems

High-fidelity speakers are quite hardy devices, but on occasion one will burn out as the result of excessive power being applied to it. Such power is not always audible. A faulty power amplifier may go into oscillation at an ultrasonic frequency, producing enough output to destroy a speaker. Perhaps this is more likely to happen with an electrostatic speaker than with a dynamic speaker, because the electrical characteristics of an electrostatic unit are more apt to throw a poorly designed amplifier into oscillation.

Another source of excessive ultrasonic energy is the tape recorder. When a recorded tape is rapidly wound forward or backward in proximity to the playback head, the great acceleration of speed converts the audible frequencies into ultrasonic ones of great amplitude, threatening damage to the speaker. Therefore, when winding a tape at high speed, make sure that volume is all the way down. For this reason some tape machines automatically disconnect their playback amplifier in the rapid-winding modes.

To protect a speaker against being overdriven by a very powerful amplifier, you can "tap down" on the amplifier's output terminals.

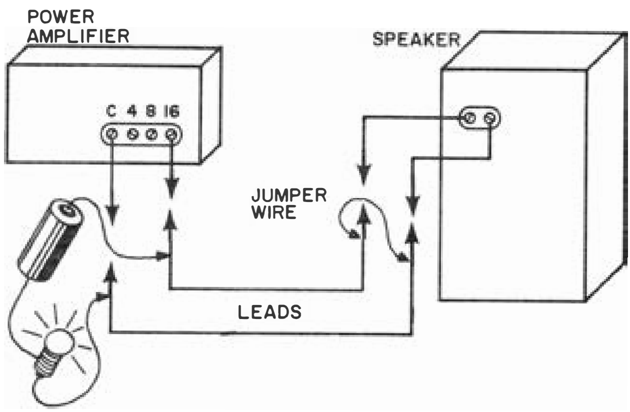


Fig. 908. *Checking for continuity of the speaker leads. Refer, also, to Fig. 804, p. 99.*

That is, you can use terminals with a lower impedance rating than the speaker's. For example, if a 16-ohm speaker is connected to the amplifier's 8-ohm tap, this cuts in half the maximum power that can be fed to the speaker. If the amplifier is capable of producing 40 watts when feeding a 16-ohm speaker from its 16-ohm tap, it can supply only 20 watts to the same speaker from its 8-ohm tap.

From time to time, a speaker is destroyed as the result of being connected to the house ac line. It happens somewhat like this: Desiring to wire several rooms of his home for sound, all to be fed from a central audio system, an audiophile installs in each room a socket into which a speaker can be plugged. And he chooses conventional ac sockets and plugs for this purpose. Inevitably, the speaker plug gets accidentally connected to the house line, resulting in immediate destruction accompanied by a violent noise. Never use conventional ac plugs and sockets for a speaker line.

To check whether a speaker is working, momentarily connect a 1.5-volt battery to its terminals (Fig. 909) . A sharp click or pop will signify that it is operating. You can test the woofer and tweeter separately by holding your ear opposite one and then opposite the other.

Absence of sound from the speaker does not necessarily mean that the speaker itself has failed. An internal connection between the speaker and the terminals on the back of the enclosure may have opened. In the case of the tweeter, partial or total loss of sound may be due to a faulty balance control. (Because tweeters are generally more efficient than woofers, the balance control is inserted in the leads to the tweeter.) This control is usually a standard part readily available from an audio supply house, and the task of replacing it in the enclosure is quite simple, requiring only a knowledge of soldering. Prior to removing the old control, of course, be sure to make a drawing of what leads go to which lugs.

With most speaker–enclosure systems, it is permissible to open up the back of the enclosure to check connections and, if necessary, replace a faulty balance control. Certain speaker systems, however, are designed so that they may be opened only by the manufacturer. Don't attempt to open the enclosure unless you are certain this is permissible.

Amplifier problems

A rather astonishing number of controls (including switches) are found in the typical preamp, particularly in a stereo unit. In addition, the power amplifier ordinarily has a number of controls. The multiplicity of controls tends to breed some confusion, particularly on the part of the neophyte, and it is not unusual to find that partial or total loss of sound is simply due to an incorrectly set control. Hence if part or all of your audio system is not functioning properly, check whether the various controls are properly adjusted.

1. Make sure that the on–off switch of the preamp is on. Frequently, but not always, this switch is combined with the volume control. Sometimes it is combined with another control or it may be completely separate. Usually the switch is actuated by a rotary or sideways motion, but on occasion it is a push-pull type, and the unfamiliar user may not understand its proper operation.

2. Make sure that the on–off switch of the power amplifier is on. Possibly you turned off this switch while checking tubes and forgot to turn it on again after putting in new ones.

3. Make sure that each of the input level sets of the preamp is turned up sufficiently. For example, if you have purchased a new phono cartridge, with better performance but substantially lower output than your old one, this may require turning up the phono level set to achieve adequate volume.

4. Make sure that you have turned up the input level set (s) of the power amplifier sufficiently. Possibly the level set was turned a good way down when using your old speaker,

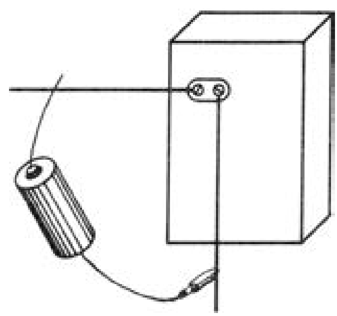


Fig. 909. *Testing the speaker for continuity.*

which was highly efficient (but otherwise a poor performer), whereas your new speaker is quite inefficient (but smooth and distortion-free).

5. The selector switch of the preamp may be turned to the wrong position. For example, you may *think* you have it turned to the FM tuner position, but actually because of dim light or the angle at which you are viewing the switch, you may have turned it to another input. If the pointer knob is attached to the shaft of the selector switch by a setscrew or by a friction fit, it is possible that the knob has slipped, giving a faulty indication of the selected source.

6. If the preamplifier has a tape-monitor switch and you wish to listen either to tape or to a source other than tape, make sure that this switch is set to the proper position. As shown

in Fig. 910, such a switch has two positions. Position 1 admits the signal from a tuner, phono cartridge, etc. Position 2 admits the playback signal from a tape machine. Position 1 is sometimes called "play" and sometimes "monitor." Position 2 is sometimes called "tape" and also some-

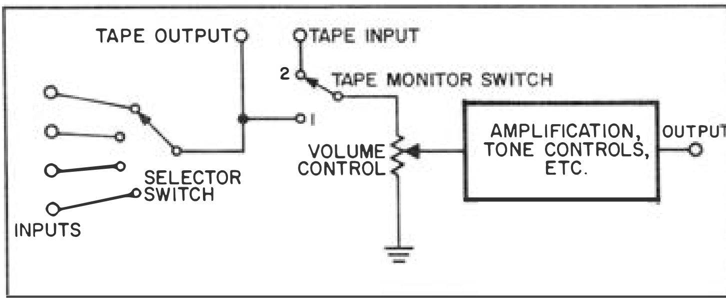


Fig. 910. *Tape-monitor switch in a preamplifier.*

times "monitor." Obviously these various terms and double usage can be confusing. Make sure that you know which is "Position 1" as identified in Fig. 910, and which is "Position 2." Refer to the preamp's instruction manual. If the tape-monitor switch is in Position 2 and you wish to listen to your FM tuner, no sound will be forthcoming.

7. Most preamplifiers have a loudness switch that converts the volume control into a loudness control, causing the latter automatically to provide an increasing amount of bass boost (and sometimes treble boost as well) as volume is reduced. The purpose of such boost is to compensate for deficiencies of human hearing as sound level is decreased. Some preamplifiers instead provide a combination of a gain control and loudness control. The purpose of the gain control (possibly labeled the volume control) is to limit the maximum sound level produced by your audio system so that it approximately corresponds to what you would originally have heard in the concert hall. Once set, the gain control is meant to be left alone, with volume thereafter to be adjusted by the loudness control, which automatically supplies bass (and perhaps treble) boost. Someone may have accidentally used the gain control to reduce volume, and the next user, relying on the loudness control, will be puzzled by the reduction in sound level.

Another possibility is that the gain control will be adjusted on the basis of a strong incoming signal from one source, say the tuner, but that this setting proves much too low for another source, say the phono cartridge. Always check the gain control setting if your preamp is one of those which incorporates a gain control–loudness control combination.

Tuner problems

Inadequate tuner output may be due to the trivial circumstance that its output level control is turned too far down. This control is sometimes mounted in an obvious position on the front panel, and sometimes it is inconspicuously mounted on the rear panel.

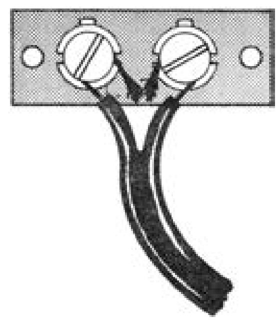


Fig. 911. *Stray strands of the FM lead-in touching each other at the antenna terminals. See also Fig. 505 on page 50.*

A more subtle cause of sound failure is the muting control found in some tuners. The purpose of this control is to eliminate tuner noise as one travels across the dial between one station and the next. However, too advanced a setting of the muting control may result in weak stations becoming completely inaudible. Even strong stations may be "squelched" if the control is turned all the way up.

Weak reception, accompanied by a good deal of noise and quite likely a good deal of distortion, may be due to a misaligned FM tuner. Tuner alignment should be entrusted only to an extremely competent technician of unquestioned reputation, preferably to the tuner manufacturer or to his authorized service agency. Weak reception may also be due to a poor antenna, to a break or short in the lead between the tuner and the antenna, or to stray strands of the lead-in wire touching each other at the tuner's antenna terminals (Fig. 911).

Phono problems

Partial or total failure of the phono system may or may not be due to failure of the cartridge. In the case of a magnetic cartridge, total failure may be due to one of the windings opening. This might happen if you were to apply a soldering iron directly to a cartridge terminal to attach a phono lead. Partial or total failure of a magnetic cartridge might be due to a partial or complete short in the winding. Piezoelectric cartridges (crystals and ceramics) tend to lose output with age, particularly the crystal cartridge. The best way of testing for a weak or dead cartridge is by substitution, either substitution of another cartridge known to be in good condition or substitution of the "live" half of a stereo cartridge for the dead half.

Little or no sound from the cartridge may be due to a short between the terminals of the cartridge; that is, between the leads attached to these terminals. Particularly in the case of a stereo cartridge, having four or five terminals, it is possible that crowded conditions inside the shell of the tone arm may cause leads to touch each other. Check the leads in the cartridge shell to make sure all are separated and that each one is properly insulated.

Sound failure may be due to an open condition or a short circuit in the leads running through the tone arm to the terminal lugs of the phonograph. You can test for an open or short circuit with your continuity checker, in a manner similar to that previously described in the section on cables.

A weak sound level, accompanied by distortion, may be due to fuzz and dirt having accumulated around the stylus, thereby restricting stylus movement.

Obviously, but sometimes overlooked, lack of sound may be due to your having made the wrong connections between the phono leads and your stereo cartridge. For example, you may have connected the leads of one of the channels to the two ground terminals of the cartridge, resulting in no sound on one channel. Inasmuch as terminal arrangement differs from one stereo cartridge to the next, take nothing for granted and follow the cartridge instructions very carefully.

Some cartridges produce considerably less signal output than others. Therefore a number of preamplifiers provide separate input jacks for low- and high-output cartridges, respectively marked "low" and "high." Weak sound may be due to a low-level cartridge being connected to the "high" input jack.

A very few magnetic cartridges produce a signal output so low that even the "low" jack doesn't provide sufficient amplification. These cartridges require "pre-preamplification," either by a special stepup transformer designed for the particular cartridge in question or by a special transistorized pre-preamplifier.

Tape recorder problems

When you can't get any playback signal from your tape machine, one of the first things to do is to check whether by misadventure you have operated the tape with its shiny rather than dull side toward the tape heads. If you haven't done such an unlikely thing, you will want to check tubes, the cable to the preamp, and the gain controls as indicated in the previous discussion. At the same

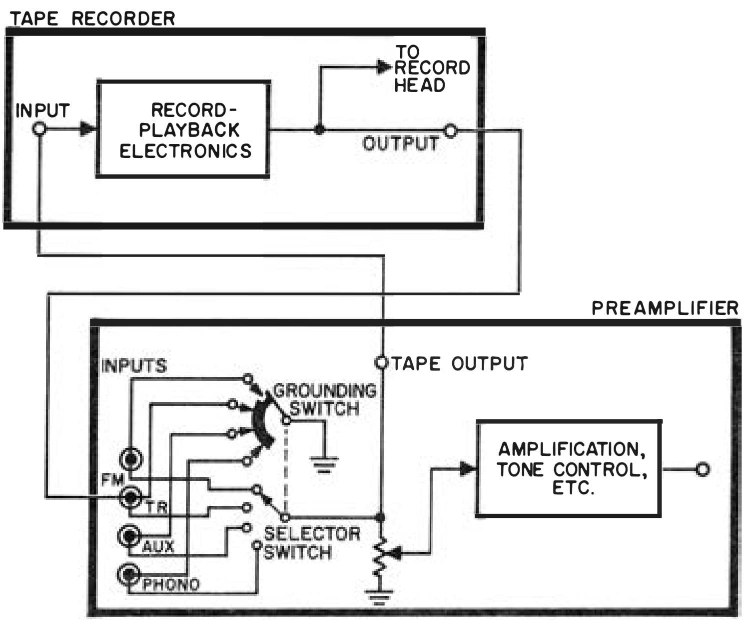


Fig. 912. *How a preamplifier may prevent a tape machine from recording.*

time, you should be alert to the possibility that the machine is malfunctioning only in the record mode (even though the magic eye or meter flickers or wiggles). You can easily find out by playing a commercial prerecorded tape or one that you recorded when the machine was operating satisfactorily. If you get a playback signal, this proves that the trouble is confined to the record mode.

Then the most obvious suspect is the oscillator tube, because a dead or weak oscillator results in a very low, and distorted, signal on the tape. On the other hand, if your tape machine has separate amplifiers for recording and playback, it is possible that some tube other than the oscillator is at fault.

In certain tape machines, when a cable is left connected between the output jack and the external preamp, the machine cannot record. Fig. 912 shows how this comes about. The type of machine in which this happens is one that uses essentially the same electronic arrangement for recording and playback, and where the machine's output jack always remains connected to this setup. The incoming signal, which is to be recorded, is therefore also routed to the output jack. Many preamps (Fig. 912) short-circuit to ground all inputs except the one in use (to prevent crosstalk between inputs). Hence the signal going through the tape recorder is shorted out by the external preamp, and little or no signal gets onto the tape.

The cure is to remove the output cable from the tape machine each time you are recording. If this isn't convenient, you will have to put together, or have a technician put together for you, an external switch such as that illustrated in Fig. 913, which can interrupt the tape machine's output signal and which can be mounted in an accessible place.

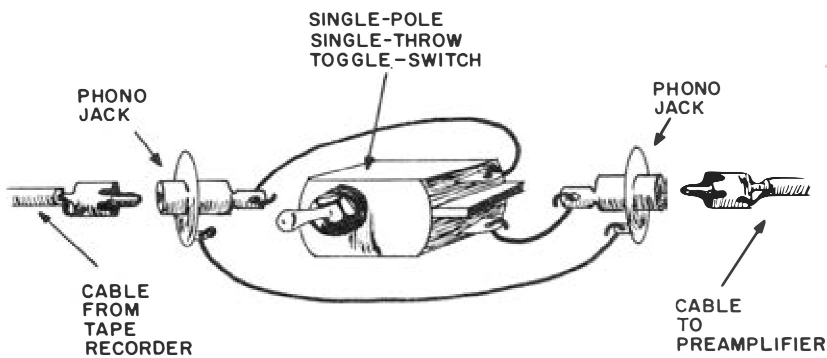


Fig. 913. *Switching arrangement for interrupting the signal from the tape recorder to the preamplifier.*

When recording from a microphone, be sure to use a microphone of sufficiently high sensitivity or you may get an unduly low level even with the recording gain control all the way up. The inexpensive microphones ordinarily furnished with home tape recorders are usually piezoelectric (crystal or ceramic) types. With age, their output diminishes, particularly in the case of the crystal microphone. For greater fidelity, audiophiles generally use dynamic microphones, or sometimes ribbon microphones. Such microphones fall into two classes in terms of output level: high impedance and low impedance. High-impedance mikes have much greater output, but suffer from the disadvantage that treble loss becomes pronounced if the microphone cable is much longer than about 12 feet. If you require a long run of cable, you will have to turn to a low-impedance microphone (some microphones can be converted to low impedance simply by turning a screw or rewiring an internal lead) . In this case you must employ a stepup transformer between the microphone and the tape recorder to feed sufficient signal into the recorder. The transformer must be located at the tape recorder end of the microphone cable.

Chapter 10
Stereo Problems

OPERATING A STEREO RATHER THAN A MONO SYSTEM IS NOT MERELY a matter of having two sound channels instead of one. Stereo introduces special problems of coordinating the two channels with respect to each other. Unless you attend carefully to these problems, it is unlikely that you will derive the full benefits of stereophonic reproduction.

Speaker phase

If the same signal is fed into both channels of a stereo system, the two speakers should be in phase, meaning that they move in unison. The cones of both speakers should be moving forward simultaneously or moving backward simultaneously, rather than one moving forward as the other moves backward. In-phase operation is important for two reasons:

1. It gives full bass response, whereas out-of-phase operation gives diminished bass response.
2. It helps focus sounds in their proper location, particularly sounds toward the center of the stage. Thus if a concert soloist occupied the center of the stage at the original performance, in-phase operation helps foster the illusion that the soloist is in the center, whereas out-of-phase operation tends to make the soloist's location somewhat vague.

Techniques of phasing the speakers of a stereo system have already been discussed at length in Chapter 6 in connection with preservation of bass response. See pages 68-70 for these techniques. Suffice it here to say that the basic method consists of feeding a low-frequency signal into both power amplifiers and reversing the leads to one of the speakers to ascertain which position of the leads yields the loudest response. That position represents in-phase operation.

In recent years, speaker manufacturers have marked their speaker terminals to indicate polarity. Thus if you are using a stereo power amplifier, or two identical mono amplifiers, a phasing problem is unlikely if you make identical connections between each power amplifier and its respective speaker (Fig. 1001). On the other hand, mistakes are occasionally made in indicating speaker

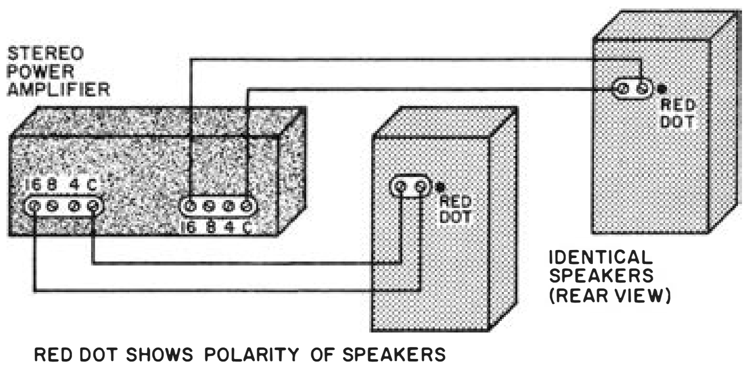


Fig. 1001. *Making identical connections to stereo speakers whose terminals are marked as to polarity.*

polarity, so that the cautious audiophile will still want to check speaker phasing. If the speaker terminals are not marked for polarity or if different mono amplifiers are used for each channel, a check of speaker phase is certainly called for.

Power amplifier phase

This is a problem only if you use power amplifiers of different brands or models for the two channels, which may be the case if you have converted from mono to stereo by adding a second amplifier. It is possible that, when the same signal is fed into both power amplifiers, the output of one will be out-of-phase with respect to the output of the other (Fig. 1002). Accordingly, it becomes necessary to check phase.

However, this does not involve you in anything new. You go through the same procedure as in checking the phase of the speakers. Again you feed the same low-frequency signal into both power

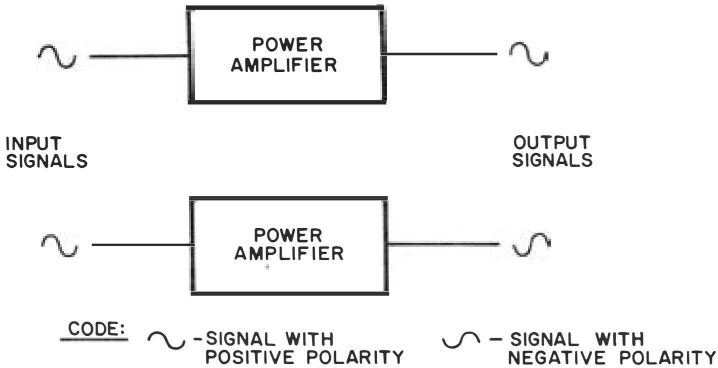


Fig. 1002. *How two power amplifiers may produce out-of-phase signals.*

amplifiers, and you reverse the leads to one of the speakers to ascertain which position of these leads yields the loudest response. As illustrated in Fig. 1003, it may be necessary to connect one speaker in opposite fashion to the other speaker to obtain in-phase operation. In Fig. 1003, it is assumed that both speakers are cor-

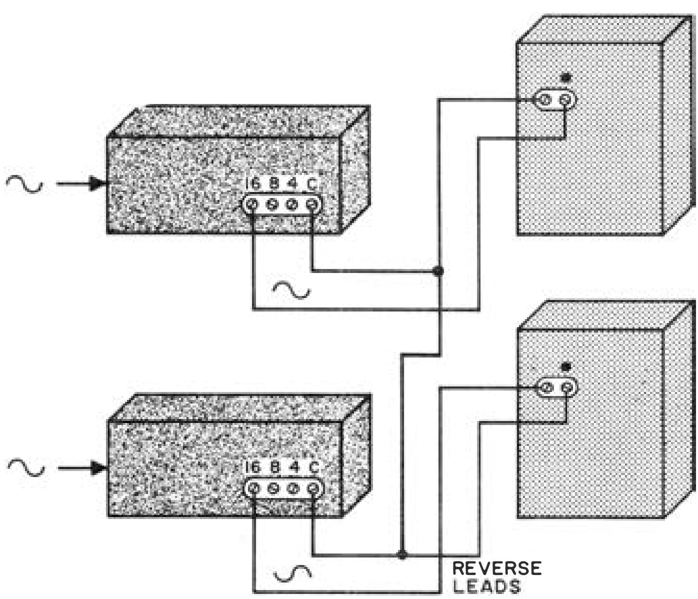


Fig. 1003. *Correcting for the out-of-phase amplifier signals by reversing the leads to one of the speakers.*

rectly marked as to polarity and that the amplifiers are out of phase with respect to each other.

Speaker and power amplifier balance

As a general rule, the best stereo effect is obtained if two stereo speakers produce equal sound levels when signals of equal magnitude are fed to their respective power amplifiers. This assumes that you are normally seated at a location roughly equidistant from the two speakers. If you normally sit much closer to one speaker than the other, it may be desirable that equal input signals to the amplifiers produce a louder response from the more distant speaker. The point is that the stereo speakers should *appear* equally loud at your customary listening position.

Although you are using a stereo power amplifier, the left channel may differ in efficiency from the right channel; that is, signals of equal level fed into both channels may produce greater output

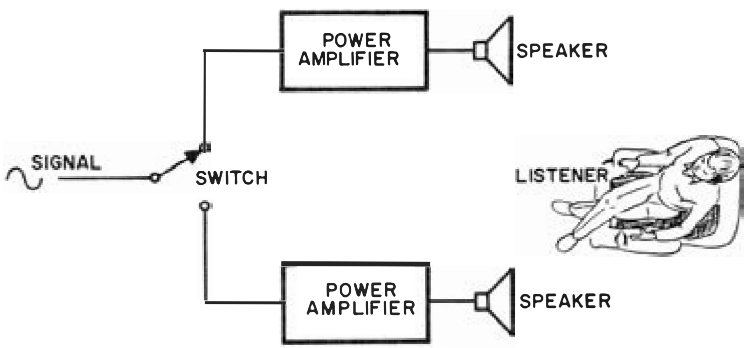


Fig. 1004. *Basic procedure for comparing the level of the left amplifier speaker with the right amplifier speaker combination.*

from one than the other. Similarly with two mono amplifiers of identical brand and model. If you are using identical stereo speakers, their efficiency may nevertheless differ slightly but significantly. And if you are using two different speakers, there is small chance that their efficiency will be the same. Here then are various additional reasons why equal signals to the left and right power amplifiers may result in different sound levels from their respective speakers. Accordingly, it is highly desirable to test for balance between the left and right amplifier–speaker combinations and to make whatever adjustment is called for.

The basic nature of this test is illustrated in Fig. 1004. With the

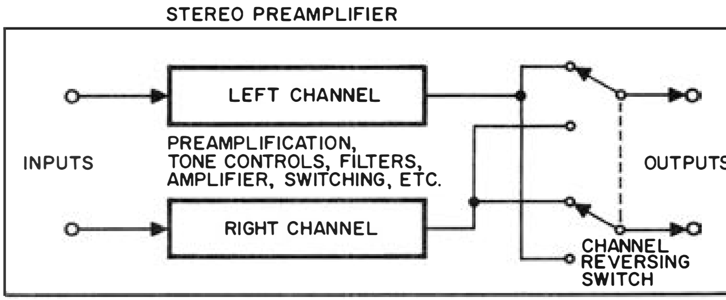


Fig. 1005. *Channel-reversing switch at the output of a stereo amplifier.*

left and right speakers in operating position and with the listener in his customary seat, the same signal is alternately fed to the left and right amplifier–speaker combinations. When the two are balanced, apparently equal sound levels should be heard from each speaker. Or, you may prefer to check balance by feeding the same signal simultaneously to both speakers and ascertaining whether the sound appears to come from midway between the speakers.

Preferably, the signal for this test should *not* be a single tone. Program material from a phono record, tape or FM station is better, especially if your left and right speakers are dissimilar.

There are various means by which you can feed a given test signal alternately to each amplifier–speaker combination. One is the channel-reversing switch commonly found in most stereo preamplifiers. This can be used, provided the switching between channels occurs at the *output* of the preamp (Fig. 1005). When reverse

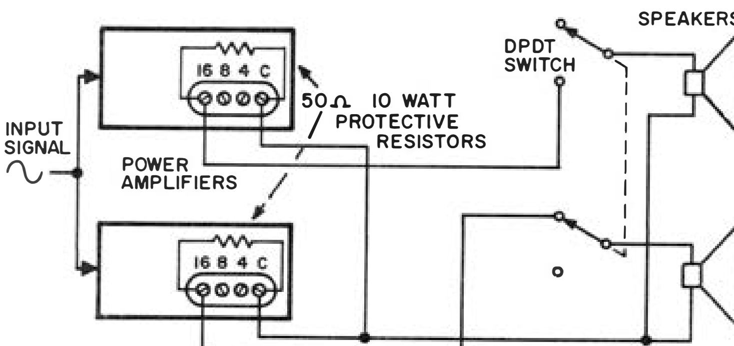


Fig. 1006. *Using a switch to connect the left and right speakers alternately to their respective amplifiers.*

switching occurs at the preamp output, this is the same as switching between channels at the input to the power amplifier (Fig. 1004).

If your preamp doesn't have a channel-reversing switch, you can follow the expedient of Fig. 1006 where the switching is performed at the output rather than the input of the power amplifier. The test signal is simultaneously fed to both power amplifiers, and the switching arrangement in Fig. 1006 permits you to connect alternately the left or right speaker to compare their output levels. Be sure each power amplifier is getting the same signal level. Some preamps contain means for checking that the output signals of their two channels are identical (a null-balance device or other means). Or it may be necessary for you to employ a Y-connector between one channel of your preamp and the two power amplifiers, as in Fig. 1007.

If you use the balancing technique of Fig. 1006, place a 10-watt 50-ohm resistor across each power amplifier during the test to pro-

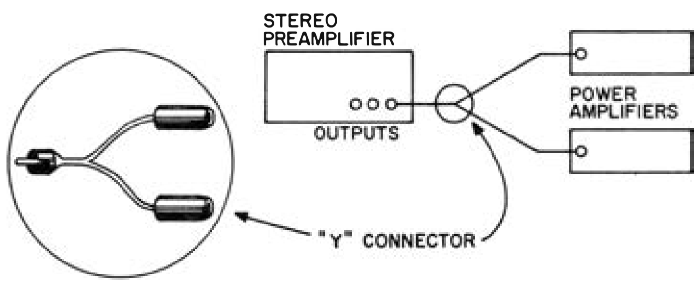


Fig. 1007. *Using a Y-connector to feed one channel of a stereo preamplifier into two power amplifiers.*

tect the output transformers. If the load across an output transformer is suddenly removed (as when disconnecting one of the speakers by means of the switch), this may create a voltage surge that can damage the transformer.

There are several ways in which you can achieve balance between the left and right amplifier-speaker combinations, should this prove necessary. Most power amplifiers contain an input level set, which will vary the amplification of a signal fed into the amplifier. You can adjust the input level set of either the left or right amplifier, as circumstances dictate, while making the test illustrated in Fig. 1004 or Fig. 1006.

If there is no level set, make the required adjustment by inserting an L-pad or T-pad between one of the amplifiers and its respective speaker, as illustrated in Fig. 1008. The pad is used with the amplifier–speaker combination that produces the louder level.

A third solution is to connect the louder speaker to an amplifier tap having a lower impedance rating than that of the speaker. Assume that the left and right speakers are both rated at 16 ohms, but that the left speaker produces more sound. Try connecting the left speaker to the 8-ohm instead of 16-ohm tap of its power amplifier, which results in a 3-db reduction in level. A 3-db de-

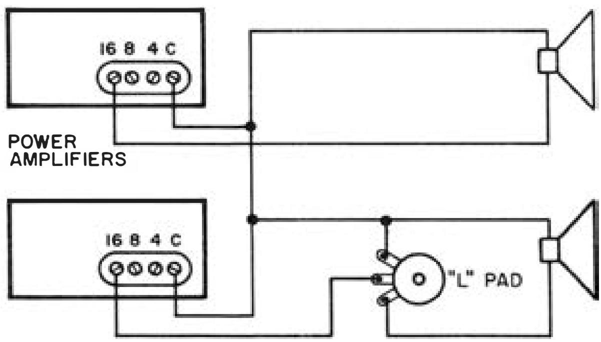


Fig. 1008. *Achieving balance by inserting an L-pad between one of the stero amplifiers and its speaker.*

crease (halving of power), is slight to the ear, but distinct. If necessary, you might even connect the left speaker to the amplifier's 4-ohm tap, resulting in another 3-db reduction.

Preamplifier balance

Now the objective is to make sure that, if equal signals are presented to the two channels of the stereo preamp, it will produce equal output signals. This involves adjustment of the balance control, or else of the relative positions of the two gain controls when the preamp has dual gain controls in place of a master gain control plus a balance control. It is quite possible that the mechanical midsetting of the balance control is not the same as the electrical midsetting, or that identical settings of the dual gain controls do not achieve perfect balance between channels.

The procedure for testing and adjusting preamp balance is this: Feed a signal from a record, tape or FM tuner to the left channel. Set the preamp's mode switch to the position that causes the same signal to go to both channels of the preamp (Fig. 1009). Adjust

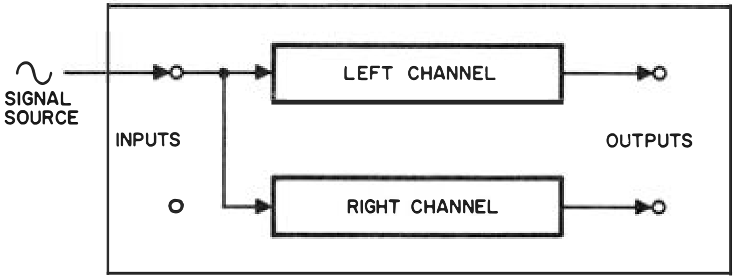


Fig. 1009. *Signal requirements for balancing a stereo amplifier.*

the balance control, or relative settings of the dual gain controls, for equal levels. It is a good idea to make this adjustment with the volume set to the listening level that you normally employ. You might also check the adjustment at several other listening levels.

To test for equal levels, you may use the technique shown in Fig. 1010, provided your preamp has a channel-reversing switch located at its output. Leave just the left amplifier–speaker combination connected to the preamp's left output jack. The channel-

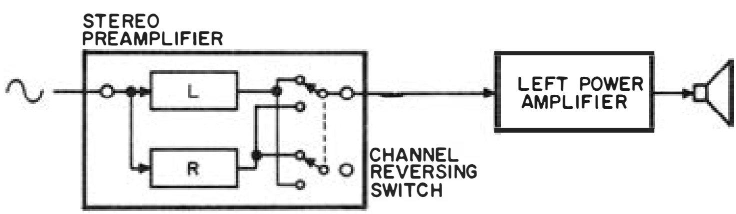


Fig. 1010. *Method of checking balance of the two channels of a stereo preamplifier.*

reversing switch enables you to alternately connect the one amplifier–speaker combination to the left and right preamp channels for testing.

An alternative testing technique is illustrated in Fig. 1011. Here each amplifier–speaker combination remains connected to its respective preamp output jack. A switching arrangement (the same as in Fig. 1006) permits you to connect one speaker while disconnecting the other. By alternating between speakers you can test whether the preamp is putting out the same signal level on each channel. This of course assumes that you have previously balanced the left amplifier–speaker combination against the right combination, as described in the preceding section.

Another means of balancing the preamp's channels is possible with certain preamps, those which have the null-balance feature. In the null-balance position the preamp delivers at each output the *difference* between the signals in each channel. When the signals are equal, the difference is zero; that is, there is no sound. Thus the balance control, or the dual gain controls, are adjusted for minimum sound.

Signal-source balance

Now we are at the point where, if equal signals are presented to the stereo preamp, equal sound levels will be produced by the stereo speakers. All that remains is to make sure that the stereo sig-

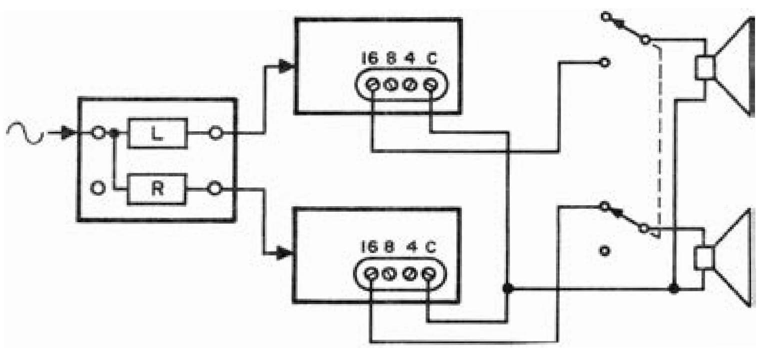


Fig. 1011. *Alternative method of checking balance of the two channels of a stereo preamplifier.*

nal source (phono cartridge, tape playback machine or FM tuner) provides equal signals to each preamp channel.

Fig. 1012 shows the basic procedure for comparing the outputs of the two sections of a stereo cartridge. Play a *mono* disc, which means that you are supplying similar signals to both channels of the preamp. The test procedure then consists of alternating

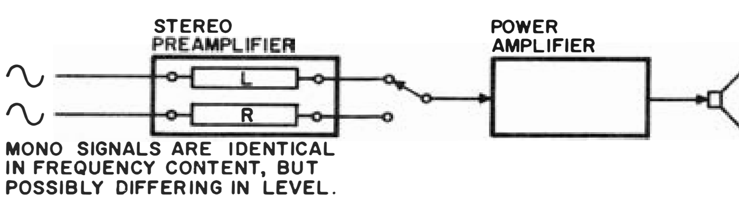


Fig. 1012. *Basic procedure for comparing the output levels of the two sections of a stereo phono cartridge.*

between the left and right channels, and thus comparing the sound levels produced by the left and right speakers.

An easy way to alternate between the left and right channels is to leave only the left amplifier–speaker combination connected to the preamp, and use the preamp's channel-reversing switch to feed the left and right signals alternately to the one amplifier–speaker combination. A second way is to use the preamp's stereo mode switch and alternate between the "mono A" and "mono B" positions, meaning that either the left signal or the right signal is fed to both channels.

A third way is to use the same speaker arrangement as in Figs. 1006 and 1011, which results in one speaker or the other being disconnected from its respective amplifier.

A fourth way is feasible with some preamps, those which have a separate selector switch (usually concentrically mounted) for each channel. Then you can turn the left selector switch to phono position while the right selector switch is turned away from phono position, and vice versa.

If the two sections of the stereo cartridge produce audibly different sound levels, the most convenient way of compensating for this difference is by the phono input level sets found in many preamps (Fig. 1013). Turn down the level set for the channel that is loudest when playing a mono disc with the stereo cartridge.

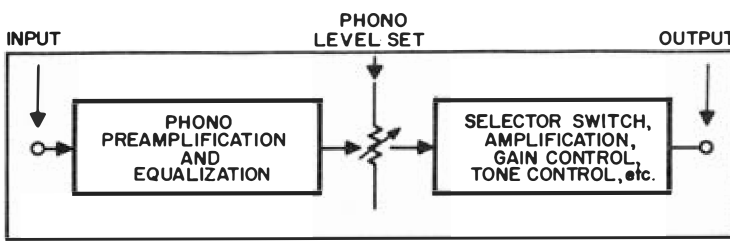


Fig. 1013. *Location of the phono level-set in a preamplifier.*

If your preamp has no level sets for the phono inputs, equality has to be achieved by means of the balance control or the dual gain controls. That is, you must take careful note of the position of the balance control, or the relative positions of the dual gain controls, that compensates for the inequality between the cartridge sections.

To check and balance the stereo signals from a tape machine, play a full-track mono tape; this could be almost any one of the test tapes available on the market. Lacking such a tape, you can test and adjust signal balance on the basis of the hiss produced by a blank tape. In any event, *don't* use a stereo tape recording. Each channel of such a recording has more or less different signal content and, even if the content were the same, the recorded level may be different on each channel.

When the tape playback signals are fed directly from the stereo tape head to the preamp, the procedure for checking and adjusting balance between the two sections of the head is the same as for a stereo cartridge (apart from the fact that now you are playing a

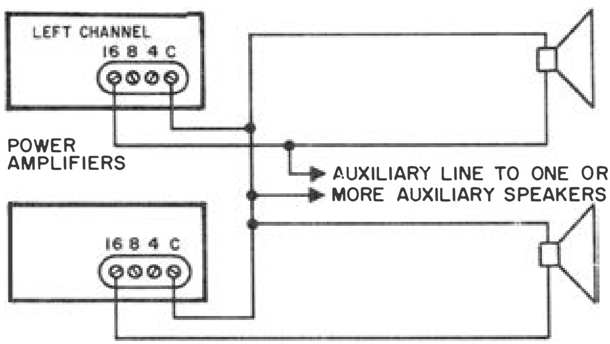


Fig. 1014. *A simple way of installing an auxiliary line to feed monophonic speakers. Note: Preamp should be set to "A + B" position when listening to auxiliary line.*

mono tape instead of mono disc) . Compensate for signal inequality either by means of the preamp's level sets for tape head input, or by the balance control or dual gain controls.

When the preamp receives the signals from the tape machine's playback amplifier, set the latter's gain controls to maximum. If the preamp has level sets for its tape amplifier inputs, use these to compensate for signal inequality. In the absence of such level sets, readjust the tape amplifier's left or right gain control, as circumstances require, to achieve signal equality. In the rare event that the tape amplifier has no gain controls, use the preamp's balance control, or dual gain controls, to compensate for signal inequality.

The procedure for testing and adjusting signal balance of the multiplex tuner is the same as for signals received from a tape amplifier. Use a mono broadcast as the program source. Adjust for signal equality with the preamp's level sets for the tuner inputs or with the tuner's left or right gain control, or with the preamp's balance control or dual gain controls.

Connecting auxiliary speakers

Audiophiles frequently like to pipe the sound from their high-fidelity system into one or more auxiliary speakers in bedrooms, dens, playrooms, patios, etc. When the source is a stereo system but the auxiliary speakers are monophonic (for example, when only one speaker instead of two is desired in the bedroom), you face the problem of how to combine the two stereo signals into a mono signal for the purpose of feeding the auxiliary speaker (or speakers).

If you plan never to use the main stereo speakers and the auxiliary speaker simultaneously, a simple way of feeding one or more auxiliary speakers, by means of an auxiliary line, is illustrated in Fig. 1014. The auxiliary line is simply connected to the left chan-

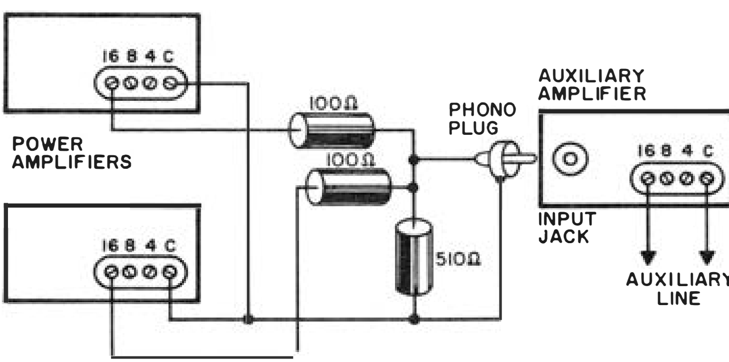


Fig. 1015. *Best way of feeding an auxiliary mono line.*

nel. When the auxiliary line is to be used, and if your program source (FM broadcast, phono disc or tape recording) is stereo, the preamp's mode switch is set to the position that combines the left and right signals, so that a mono signal goes out on this line.

However, you may want to operate an auxiliary mono speaker while the main speakers are reproducing stereo. There are several methods of doing this. The best is illustrated in Fig. 1015. The left and right signals are combined, via the resistor network shown, and fed to a third power amplifier, which supplies the auxiliary line. The input level set of the third power amplifier is adjusted to provide the desired amount of amplification of the auxiliary signal. If this amplifier has no input level set, a 500-ohm potentiometer can be substituted for the 510-ohm resistor in Fig. 1015.

A second method of operating the auxiliary line simultaneously with the stereo speakers is illustrated in Fig. 1016. Here the auxiliary line is directly connected between the "hot" terminals of each power amplifier, say between the two 8-ohm taps. In this situation you must reverse the phase of the signal going to one of the power amplifiers, which can be done by means of the phase-reversal switch

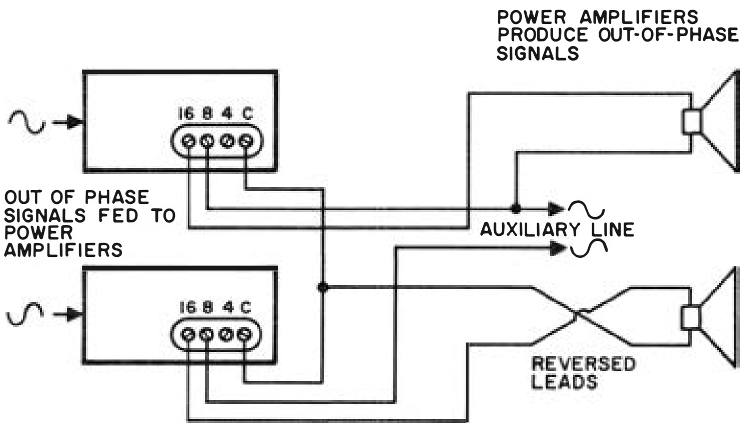


Fig. 1016. *Alternative method of feeding an auxiliary mono line. Note: For less reduction of stereo separation of main speakers, connect the auxiliary line to different output taps than the main speakers.*

contained in many stereo preamplifiers. Thus the 8-ohm terminal of one power amplifier is negative when the 8-ohm terminal of the other amplifier is positive. Accordingly, the auxiliary line receives a sum signal. (The auxiliary line signal is the *difference* between the signals at each 8-ohm tap when this line is connected as in Fig. 1016. The *difference* between a positive and a negative signal is the sum of the two signals. To illustrate, assume that the voltage at the left amplifier's 8-ohm tap is +4 when the voltage at the right amplifier's 8-ohm tap is −4. The difference between +4 −4 is 4 − (−4), which is +4 +4 = 8 volts.)

Note in Fig. 1016 that since the output signals of the two amplifiers are out-of-phase, it is necessary to reverse the leads between one of these amplifiers and its speaker, thereby restoring in-phase operation of the two speakers.

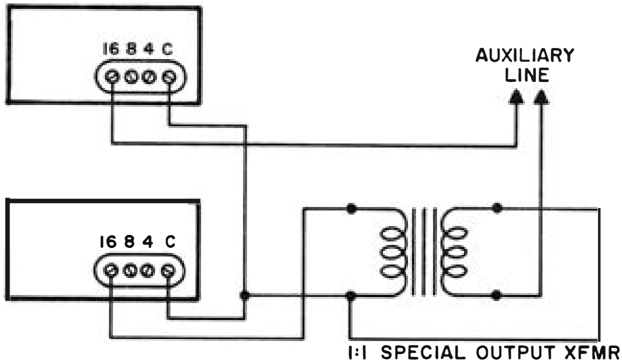


Fig. 1017. *Using a special output transformer to reverse phase of one channel and provide isolation between channels when feeding an auxiliary mono line.*

An alternative approach, which also doesn't require a third power amplifier, is shown in Fig. 1017. Here a special output transformer is used to reverse the phase of one of the channels.

The techniques of feeding a mono auxiliary line represented in Figs. 1015, 1016, and 1017 can be used equally well to feed a center speaker in a stereo installation. Sometimes a center speaker is desirable when the left and right hand speakers are spaced so far apart as to form a very wide angle with respect to the listener.

Chapter 11
Kit-Building Problems

KIT-BUILDING IS A WIDESPREAD ACTIVITY TODAY. NOT ONLY IS IT A means of acquiring audio components (as well as test instruments, amateur radio equipment, marine devices, etc.) at substantial savings, but it is educational and enjoyable in itself. When the kit is finished and put into operation, it is an exhilarating experience to find that from your hands has sprung an electronic device that works perfectly.

But when the assembled kit works imperfectly or not at all, you are apt to find yourself deeply frustrated. Fortunately, every reputable kit manufacturer guarantees that he will repair the kit at a reasonable fixed fee. Still, you are confronted with an extra cost plus a delay before the kit is returned. So it may be safely assumed that many an audiophile is interested in knowing what he can do to maximize his chances of success in assembling a kit.

What to build first

It makes elementary good sense that the first kit you build should be the simplest. The experience and knowledge acquired in your initial effort, plus the sense of confidence gained, can be very helpful in putting together more complex kits later on.

This does not imply that certain kits are difficult to build. It is only that some take more steps than others. Taken one at a time, each of the carefully programmed steps in a modern kit is simple and brief.

A power amplifier is usually the easiest component to assemble.

Some such kits require as little as 3 or 4 hours of work. Next you might turn to the preamp. This generally requires substantially more time, but that is about all there is to building a preamplifier compared with a power amplifier—more time.

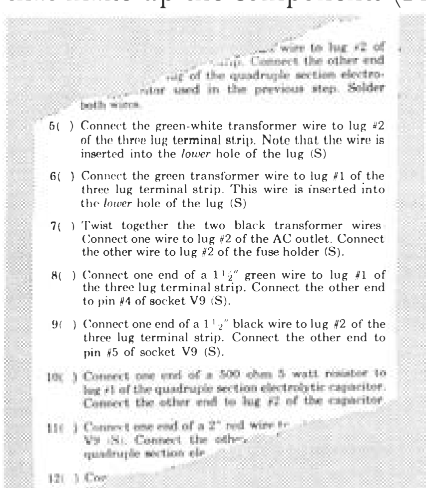
Third in the desirable order of construction is the FM tuner. Its complexity is of about the same order as that of a preamplifier, but it may require somewhat more care in assembling and arranging parts exactly as specified by the manufacturer. Moreover, the FM tuner may require alignment. If so, the kit manufacturer will provide all the means for doing this yourself, without the need for test equipment.

A component such as a tape recorder probably should be left for last. Electronically, it is as complex as any of the others, and in addition it may present the task of assembling the transport mechanism.

Getting started

When a kit arrives in the house, you are apt to be impatient to get started on it as soon as possible. Curb this impatience.

The first step is to read all the introductory remarks in the instruction manual, up to the point where the step-by-step instructions begin. From this you will gain a helpful perspective on the job ahead, valuable pointers that will facilitate your task, warnings against possible pitfalls, and acquaintance with the mechanical and electronic parts that make up the components (Fig. 1101).


wire to lug #2 of ... Connect the other end ... of the quadruple section electro- ... used in the previous step. Solder both wires.

5( ) Connect the green-white transformer wire to lug #2 of the three lug terminal strip. Note that the wire is inserted into the *lower* hole of the lug (S)

6( ) Connect the green transformer wire to lug #1 of the three lug terminal strip. This wire is inserted into the *lower* hole of the lug (S)

7( ) Twist together the two black transformer wires Connect one wire to lug #2 of the AC outlet. Connect the other wire to lug #2 of the fuse holder (S).

8( ) Connect one end of a 1½" green wire to lug #1 of the three lug terminal strip. Connect the other end to pin #4 of socket V9 (S).

9( ) Connect one end of a 1½" black wire to lug #2 of the three lug terminal strip. Connect the other end to pin #5 of socket V9 (S).

10( ) Connect one end of a 500 ohm 5 watt resistor to lug #1 of the quadruple section electrolytic capacitor. Connect the other end to lug #2 of the capacitor.

11( ) Connect one end of a 2" red wire to ... V9 (S). Connect the other ... quadruple section ele...

12( ) Con...

Fig. 1101. *Excerpt from a kit instruction manual.*

The next step is to make a detailed check of the parts list, to find out if anything is missing or faulty. Check resistors, not only for value as indicated by colored stripes (the color code is usually given in the manufacturer's manual), but also for wattage rating as denoted by resistor size (shown in the manual). Check capacitors not only for value but also for voltage rating. Although the construction of some kits is divided into stages, with parts grouped according to stage, it is advisable to check the parts list for *all* stages before commencing any work. If something is missing or imperfect, you can immediately write to the manufacturer for replacement (Fig. 1102).


All resistors are ½ watt 10%

Fig. 1102. *Excerpt from a kit parts list.*

In many kits the parts come in fairly helter-skelter fashion in bags, boxes and envelopes. Even before checking the parts list, arrange everything systematically, by type and by value. For example, separate the screws, nuts and washers from each other, and sort according to size. Arrange resistors from lowest to highest in value; also the capacitors. And so forth. Checking the parts list will then be a snap. And assembly of the kit will go much quicker.

Faulty parts are seldom encountered in a kit. Still, if you have access to the necessary equipment, it is worth taking some extra time to check resistors and capacitors. This requires an ohmmeter and a capacitor checker (which can themselves be built from inexpensive kits). Check resistors for value, subject to the tolerance limits indicated by the markings on them. Check capacitors for value, subject to their marked tolerance limits, and for leakage.

Proper tools

A few, relatively inexpensive, but appropriate tools are required for kit building. The instruction manual that comes with your kit will usually tell you what is needed. Typically, these tools consist of a couple of screwdrivers of different sizes, a small low-wattage

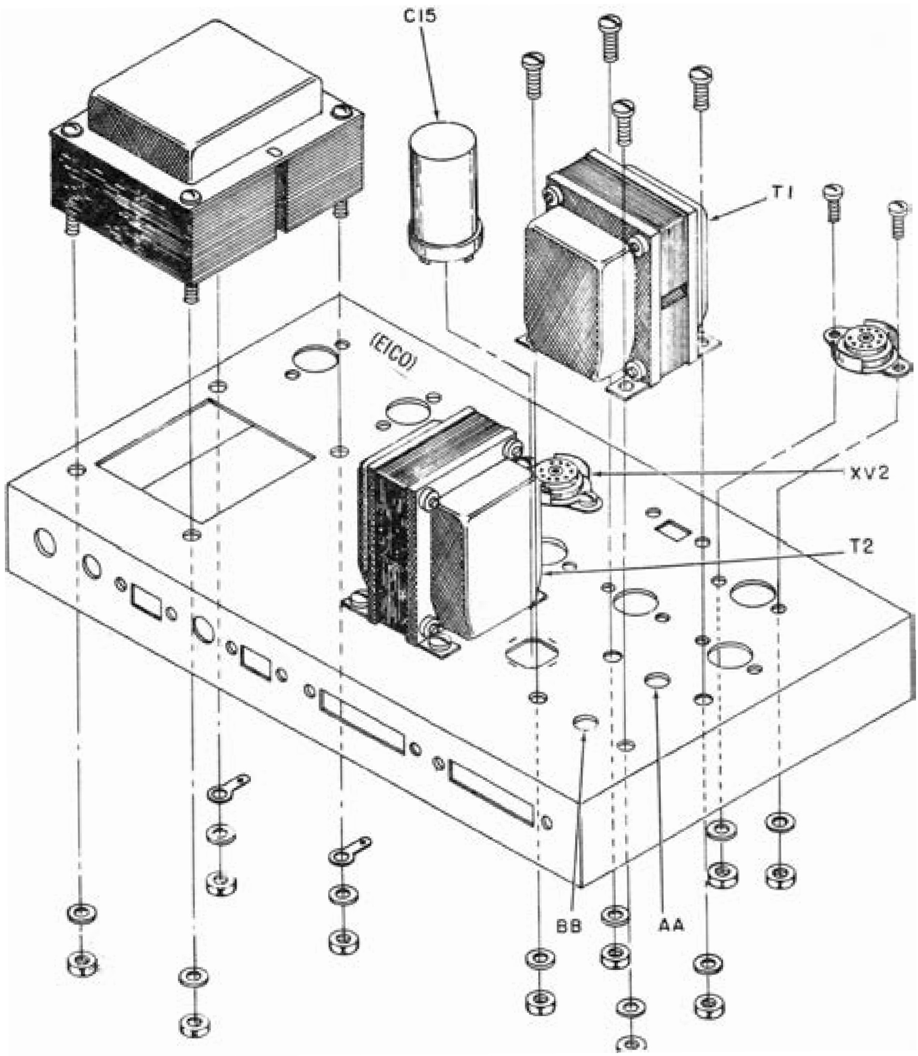


Fig. 1103. *Exploded view shows positioning of parts.*

soldering iron (not over 75 watts and preferably lower), a long-nose pliers, a regular small pliers, a wire cutter and stripper, and a side cutter. (You will probably find that these items are also useful for other things than kit building!)

Equip yourself with the proper tools before starting work on the kit. It is discouraging to be balked in the middle of things because you don't have the necessary tool. Worse, when you're impatient, there is a tendency to resort to makeshift tools, which increases the possibility of faulty assembly or damage to a part.

Preliminary precautions

In assembling a kit, you are heavily dependent on pictorial diagrams. Study each one carefully to make sure that you have the correct visual perspective. That is, be sure you can differentiate correctly between the front and back of the chassis, between bottom

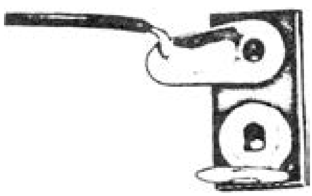


Fig. 1104. *A good solder connection.*

and top, between left and right. Every so often a kit builder assembles things "inside out." He may mount everything on one side of the chassis that should have been mounted on the other side. This

Fig. 1105. *A poor solder connection (cold solder joint).*

is quite likely to happen when the component uses a vertical chassis (the tubes lie horizontally). Here he may make the mistake of mounting on the front of the chassis the components that are meant to go on the back, and vice versa (Fig. 1103).

A similar mistake is that of mounting tube sockets from above the chassis rather than from below. Mounting the socket from above may short-circuit the socket lugs against the chassis.

Every electronic component includes one or more transformers. Sometimes the transformer leads are already cut to size, stripped and ready for soldering. At other times you have to do your own cutting and stripping. In addition, transformer leads are sometimes covered with enamel, which has to be carefully scraped off to permit a good electrical connection. Failure to fully remove the enamel means trouble.

The soldering problem

Probably more kit troubles are ascribable to poor soldering than to all other causes combined. While soldering is an art, it is a simple one in which you can become adequately skillful within half an hour. Before beginning your first kit, practice soldering for a while with some scrap pieces of wire.

A good solder connection is smooth and shiny in appearance, as shown in Fig. 1104. A poor connection, called a cold solder joint, is dull, rough and granular (Fig. 1105). A cold solder joint is usually due to insufficient heat or to movement of the "work"—the leads being soldered—before the solder has cooled. It generally means a poor electrical connection and therefore the likelihood of malfunction.

Solder isn't like glue. You don't just melt it and let it pour around the leads to be joined. The essence of good soldering is that the

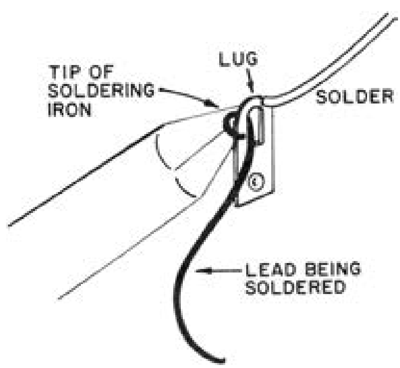


Fig. 1106. *Correct soldering technique.*

surfaces to be joined must be hot enough to melt the solder; only then does the solder achieve a secure electrical and physical connection. Therefore the technique is to apply both the solder and the tip of the soldering iron to the work, without the solder and tip touching each other (Fig. 1106). When the "work" gets sufficiently hot, it melts the solder, which flows and creates a good bond. You can improve your chances of a sound joint by "pretinning"—by applying solder beforehand to each lead to be attached to a lug.

An iron rated between 25 and 75 watts is usually suitable for audio kits. A lower-wattage iron, being smaller, can get into tight spots more easily. But an iron of higher wattage is desirable when a connection has to be made to a ground lug—one connected to chassis—because the chassis drains the iron of its heat. A higher-wattage iron is also desirable when soldering a large number of leads to one lug. Some soldering irons have removable tips that come in various wattages. One such iron and a couple of different tips can meet your needs.

If you are using an iron of low wattage to get into a tight spot, and are having trouble getting the work hot enough to melt the solder, let the iron rest a minute or two to regain heat. Then try again.

Too much heat creates problems. Solder will not flow and otherwise act properly if subjected to excessive heat. An over-hot iron can destroy a transistor. It can melt the insulation around the inner lead of a shielded cable, causing a short circuit between this lead and the shield. Too hot an iron can also burn out a small transformer, such as the i.f. (intermediate frequency) transformer

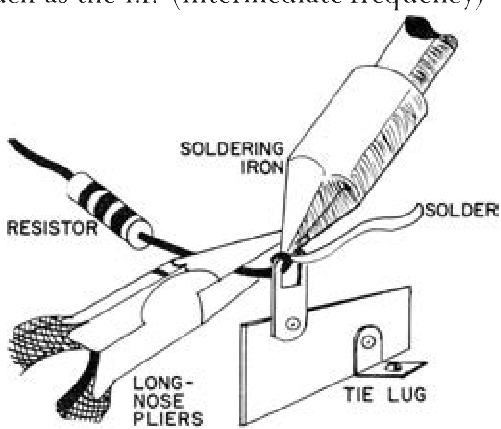


Fig. 1107. *Using a long-nose pliers as a heat sink.*

in a tuner. It can damage capacitors. It can cause resistors to change value and become noisy.

To be on the safe side when using a relatively high-wattage iron, use a heat sink—a metal object which draws away the iron's heat before it reaches the part in danger. A heat sink can simply be pliers held between the iron and this part (Fig. 1107). Since three hands are seemingly called for in Fig. 1107, you can make yourself a self-clamping heat sink instead of using pliers. This can be fashioned out of an alligator clip to which a piece of heavy wire has been soldered (Fig. 1108).

Acid-core solder is absolutely taboo for electronic kits. Connections made with this solder will eventually corrode, resulting in

Fig. 1108. *A self-clamping heat sink.*

poor electrical contact. The only kind of solder you may use for audio kits is the rosin-core type. Solder is a combination of tin and lead. The best combination is about 60% tin and 40% lead. Such 60-40 solder has the lowest melting point and sets most quickly, minimizing the danger of cold solder joints. The 60-40 is the most expensive solder, but worth the difference. Other acceptable combinations, costing less, are 50-50 solder (half tin, half lead), and 40-60 (40% tin, 60% lead).

Before being used the iron must be tinned; that is, its tip must be covered with shiny solder. Keep the tip clean as you work. Rub it occasionally on a heavy piece of aluminum foil, or on white or kraft paper, or a piece of scrap cloth. Retin the tip as necessary. You may even have to file the tip smooth.

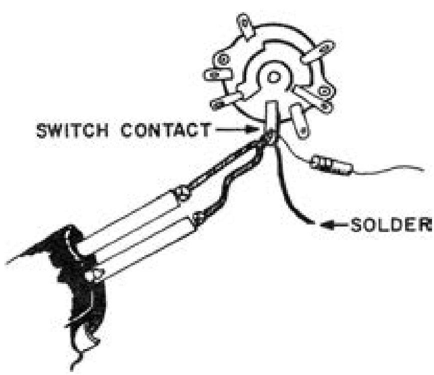


Fig. 1109. *Positioning a switch lug downward while soldering so that solder won't flow into the switch contact.*

Don't use too much solder. Let just enough flow to fill the lug that holds the leads being soldered. If too much solder flows onto the work, reheat the work and remove excess solder with the tip of a screwdriver.

When soldering leads to a switch, be extra careful not to let solder flow into the switch contacts. Solder in its contacts will make the switch inoperative. One safeguard is to position the switch so that the lug being soldered points downward (Fig. 1109)—the solder is unlikely to flow upward into the contact. If solder should get into the contact, reheat the lug until the solder in the contact melts. Then, with the switch in your hand, rap either the switch or your hand sharply against your worktable, and the solder will fly out. *Guard your eyes while doing this.*

After soldering a connection, let it cool for several seconds before touching any of the parts involved. Otherwise you may get a cold solder joint. If you do, reheat the connection, remove some of the solder with a screwdriver tip, and apply fresh solder.

Lead dress

Lead dress is the arrangement of leads, cables and parts. The instructions and pictorials of the manual describe exactly how every part and lead should be arranged. Often, to achieve correct

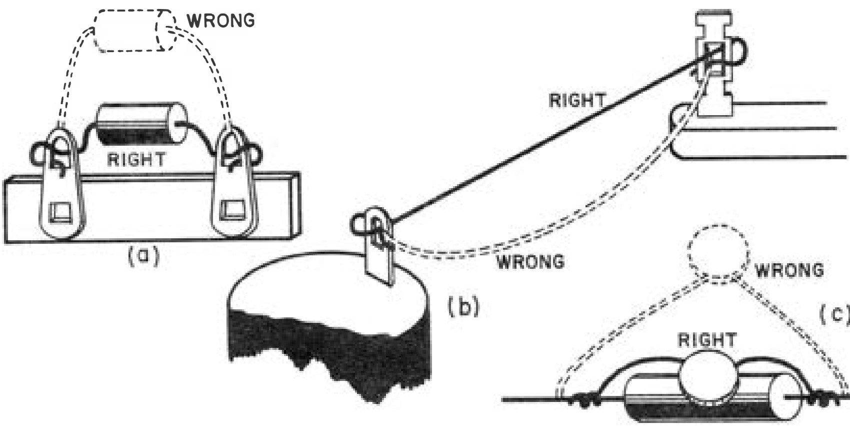


Fig. 1110. *Right and wrong ways of keeping leads short.*

operation, very short leads are called for. Fig. 1110 shows the right and wrong ways of dressing leads when short ones are required.

Frequently, you are told to dress a lead along a route that is not the shortest nor the most conducive to neat appearance. Do *exactly* as the manual says. There is probably a very good reason for the designated route, such as avoiding hum pickup, crosstalk, oscillation or other problems. The manufacturer may have invested hours of resarch in determining the best position of a lead. In sensitive stages, such as the magnetic phono section of a preamplifier, changing the location of a lead by a small fraction of an inch can make a significant difference in operation. At times you will be told to dress a resistor, capacitor or other part tightly against the chassis. Do so. If the part and its leads hover in the air, they may act like little antennas, picking up hum and other troubles.

Don't neglect to use "spaghetti" (insulated sleeving) according to instructions. Otherwise you invite short circuits and other damaging electrical contacts.

Physical connection of leads to lugs

Unless a kit uses printed-circuit boards, almost all soldering connections are made to the lugs of tube sockets, filter capacitor sockets, terminal strips, etc. Kit manuals usually instruct that, prior to soldering, you should make a secure physical (mechanical) connection between the lead and the lug by wrapping the lead pigtail-fashion around the lug and crimping the lead with a pliers.

There is danger in taking these instructions too literally. While building the kit you may have to unsolder a lead to correct a mistake. Or at some time in the future you may have to replace a part. If the lead of this part is firmly wrapped around a lug, you may have a wrestling match in prying the two apart, possibly resulting in damage to the lug, the part in question or some other part.

The preferable practice is to make the physical connection just secure enough to hold the lead in place (Fig. 1111). A solder con-

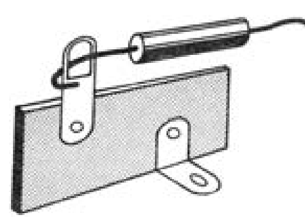


Fig. 1111. *Making a loose physical connection between a lead and a lug to facilitate removal of the lead.*

nection that is electrically sound will also be physically sound. (On the other hand, a connection that is physically strong may not be electrically sound.) Support for this point of view has appeared in several quarters.[1] I have assembled many kits, always

[1]For example, in the September 1956 issue of *Electrical Manufacturing* there appeared an article, "Reliable Soldered Connections Without Mechanical Joints," by J. Roy Smith, head of the Reliability and Standards Branch, United States Navy Electronics Laboratory, San Diego, Calif.

relying on the solder connection and using minimal physical connection between the lead and the lug and I have never encountered trouble with such a connection.

Check your work

All humans make errors, and, if you build a kit involving 100, 200 or even 300 steps, at least one error is almost inevitable. Hence it is important that you check your work systematically and frequently. A good procedure is to stop after every 10 assembly steps and retrace these steps. Check whether you have installed the right part, whether each lead is connected to the correct place, and whether the solder connection (if any) is bright and shiny and completely fills the hole of the lug. If there are several leads in a lug and the lug isn't completely filled with solder, one of the leads may be loose or at least not making a sound electrical connection.

Some kit builders who own an ohmmeter take the added precaution of checking whether they have a good electrical connection (Fig. 1112). Thus, if a lead is wired between points A and B in

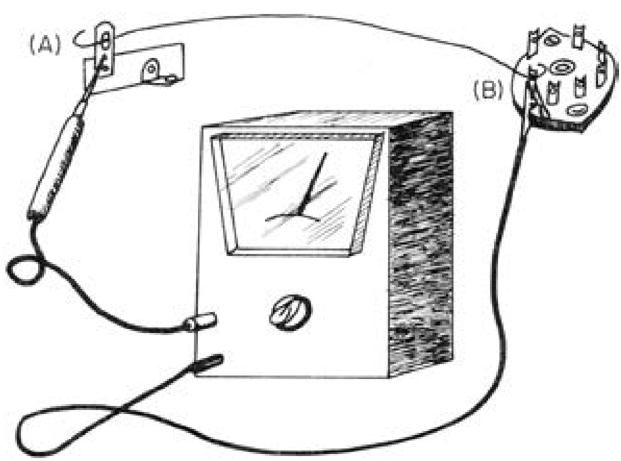


Fig. 1112. *Using an ohmmeter to test for a good electrical connection.*

Fig. 1112, they check for zero resistance, which would show there is a good electrical path between points A and B. Alternatively, you can use your home-made continuity checker, as illustrated in Fig. 1113.

Haste makes waste

The proficient kit builder sometimes grows impatient with the deliberate step-by-step pace of the instruction manual and undertakes what he considers time-saving measures. Thus he may install a part or solder a lead before the instructions say to do so. Mistakes are the usual consequence.

Stay with the instructions. Although the assembly procedure is finely divided into many small steps, do not try to save time by consolidating groups of steps. After all, such consolidation will not reduce the number of parts to be installed and the number of connections to be made and soldered.

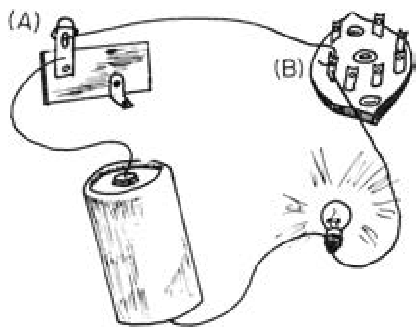


Fig. 1113. *Using the continuity checker to test for a good electrical connection.*

Some kit builders become impatient and try to rush the work to a quick conclusion. This results in a progressively sloppier assembly and leads to outright mistakes or to poor connections that cause malfunction. Kit building is not intended as a race against time. There is no more point in racing through a kit than through a meal prepared by a master chef. Adapt the pace of the work to your temperament and spare time. When you start getting tired, stop. Resume the work an evening or two later, only when you again have the energy and patience to do your best work.

Trouble after completion

Despite your best efforts, the assembled component may not work on completion. First check the obvious things. If the component has an on–off switch, make sure that it is in the on position. Make sure that the power plug is inserted firmly into the house wall socket. Make sure that this socket is alive by plugging in a lamp, radio or other appliance. Be sure that you have firmly inserted all the tubes into the component. Make certain that each tube has gone into the correct socket. Check whether you have inserted the fuse (if any). Visually check the fuse to see whether it is good. (Or use your continuity checker to tell whether the fuse is good.) In the case of an FM tuner, have you connected an antenna? What about connecting cables between components? Etc.

If none of this helps, carefully read your instruction manual for trouble shooting hints. Next, review all your construction steps, looking for cold solder joints, connections that somehow you forgot to solder, resistors and capacitors of wrong value, leads connected to the wrong places, parts omitted, transformer color-coded leads mixed up, and so on. Since we all have blind spots, it is human in reviewing your own work to make the same mistake twice or more. If you can get a friend to check your work, he may find in a matter of minutes an error that might elude you for hours.

After all this, if necessary, and if you have access to a vacuum-tube voltmeter or a volt-ohm-milliameter, you might check resistances and voltages at each pin of the tubes (or at each terminal of transistors). *Resistances are checked with the power off;* voltages with the power on. The instruction manual usually provides resistance and voltage charts showing the readings that should be obtained—within certain tolerance limits, usually ±20%—between ground and each tube pin. If these charts aren't in the manual, write to the manufacturer for them. When you find a voltage or resistance reading well outside the proper range, check all your work again in the vicinity of the tube where you obtained the reading.

If you still haven't found the trouble, your only recourse is to the manufacturer or to one of his authorized service agencies. (The manufacturer will supply a list of such agencies.) Resist the temptation to take the kit to your local service technician. This is not intended as a slight on the local technician. However, he is accustomed to repairing equipment that was originally in working condition, whereas your assembled kit is outside this category. He may have to spend hours, *at your expense,* tracking down your blunder by comparing your work with the circuit diagram in the kit's manual. But the manufacturer's technicians are thoroughly versed in the circuitry of your kit and in the mistakes that a kit builder is apt to make. They can generally spot the difficulty in minutes rather than hours. In any case, however much time they spend, you know that you only have to pay a moderate, fixed fee.

Before sending your kit to the manufacturer, write a clear, brief, orderly, legible letter describing the symptoms. Possibly the manufacturer will answer with a suggestion that saves the day. If not, pack the kit carefully. Place it in a carton a few inches bigger than the kit, with crumpled newspaper between each wall of the carton and the kit. (Save the original carton for such possible use.) Fasten and tie the carton securely. Railway Express is usually the best way to ship the kit. While more expensive than parcel post, it tends to give greater assurance that the kit will arrive intact. Promote the chance of the carton arriving intact by clearly writing "fragile" on each surface.

PRINTED IN THE UNITED STATES OF AMERICA

Hi-Fi Troubles

..how you can avoid them
..how you can cure them

Herman Burstein

No matter how ambitious or costly your audio equipment may be, its performance cannot be truly described as hi-fi unless it is troublefree, for even the slightest fault is bound to mar the perfect reproduction one expects from modern hi-fi installations.

This book tells you in the simplest possible terms how you can avoid hi-fi troubles and, if they occur, how you can cure them. An abudance of clear illustrations help to make the very lucid text even easier to understand.

The first chapter explains why we have audio troubles and subsequent chapters deal with tools of the trade, the art of substitution, hum problems, noise problems, bass and treble problems, distortion, no sound or weak sound, stereo problems and kit-building problems.

In home-constructed gear, the first essential is to avoid building-in trouble through using components with a actual or potential faults. This book describes how to pre-test components so as to ensure that every part is in good working order when it is assembled into the complete equipment.

It explains how to minimize chances of troubles with audio equipment of many different kinds. There are instructions on making and using test equipment; how to use your tuner as a substitute signal-source; and how to pinpoint trouble down to one component.

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