gain is that of defective coupling capacitors.

Still another source of trouble can be a cold-solder joint. Such joints seem good at the time they are made and indeed, they may work properly for some time, but ultimately some resin will penetrate among the various leads making up the connection and this will cause the resistance to rise, sometimes to infinity.

Another possibility is that one half of the output transformer has opened. This can lead to both loss of gain and to instability, especially when the output stage derives its bias through a dropping resistor in the cathode circuit.

Another possibility is that the feedback-loop resistor has changed value. If it has become smaller, more and more voltage will be fed back from the output stage, thereby reducing the gain of the amplifier. Further, excessive feedback can cause instability because of shifts in phase of some of the components, especially the output transformer. Although there are always phase shifts, they are not always great enough in their effects until the feedback increases beyond that intended by the designer of the equipment.

If feedback capacitors open or become larger, depending upon their location in the circuit, instability can arise because of excessive feedback or by additional phase shifts which the capacitor was designed to counteract.

If you have an AC VTVM, check the gain of each stage and find the one which is causing the trouble; then concentrate your search there. It may be helpful to disconnect the feedback circuit, lest it influence the gain. If all stages operate normally, you must then look into the feedback circuit. Measure the gain at various frequencies with and without feedback; if it is reduced when feedback is applied by more than 20 or 25 db, than excessive feedback is probably present. For an accurate appraisal of the feedback problem, consult the design notes of the equipment to see just how much feedback is supposed to be present.

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Low Gain of Power Amplifiers

Q. Thanks for your reply to my recent letter. I have another problem which has plagued me for some time now. Several years ago I built a 20-watt Ultra Linear amplifier using two 807's, two 6SN7's and a 5U4. It has served me faithfully for some time but it has developed troubles which I cannot locate. The gain has fallen off considerably, and new tubes did not help. Voltages all check normal. I replaced the filter capacitors because, with my input level control rotated all the way to the left, motorboating was audible. The replacement of these capacitors did not help. I would appreciate hearing from you concerning this matter with any advice you can give. James O. Valestin, St. Louis, Mo.

A. The first thing to check is the cathode bypassing. Failure of such a component will result in a reduction in gain and perhaps some instability. A drop in gain accompanied by an even greater reduction in bass would be further evidence that a cathode bypass capacitor is open.

Sometimes the amplifier will behave abnormally if there is an open grid resistor present. Under this condition, the grid is charged excessively with electrons from the cathode of the tube, thereby cutting itself off.

Another possible source of the loss in

Crackling in Amplifiers

Q. I have on hand an old 50-watt PA amplifier. The tube lineup consists of two 6J7's, two 6N7's, four 6L6's, one 6X5, and two 5V4's. The nature of the trouble is a crackling noise in the output. I have replaced many resistors and electrolytics with no improvement. J. L. Cosette, Quebec, Canada.

A. First, check the tubes. This is always the first thing to do when servicing equipment, partly because they can cause a multiplicity of troubles, and partly because they are the easiest to check, especially when you have replacements. If tubes check normal, look to the coupling capacitors. Since your unit is an old one, it probably contains many waxed paper capacitors, and such units often give rise to the type of trouble you described.

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Hiss Level in Preamplifiers

Q. How can I reduce the background tube noise which occurs when the preamplifier is in the phonograph position? The noise shows up especially when solo instruments are playing. C. I. Schup, Lawndale, Calif.

A. Perhaps the background noise you notice is the result of running your power amplifier at excessively high level compared to the level of the preamplifier. In some instances, the stages following the preamplifier volume control have considerable noise content. If the gain of the power amplifier is set too high, too much of this noise will get into the amplifier. Simply reduce the gain of the power amplifier and increase that of the preamplifier.

Unfortunately some power amplifiers are not equipped with input gain controls. In such cases, you may find it advisable to modify your unit to include such a control. If, for some reason, you are unable to include the control, make up a voltage divider of fixed resistances.

Sometimes the noise results from poor signal-to-noise ratio in the phono stage of the preamplifier. The signal-to-noise ratio becomes worse as pickups with smaller and smaller outputs are used. If the trouble is in the phono stage, you can determine it by raising and lowering the volume of the preamplifier. If the hiss level changes as the control is rotated, the trouble lies in the phonograph stage. There is probably little you can do about this trouble except to use a cartridge whose output is higher than that of your present cartridge. Before doing this, however, check to see if the manufacturer of your present cartridge has a stepup transformer designed to work into preamplifiers requiring higher input drive.

If you are handy with a soldering iron, you can try replacing some of the resistances in the phono stage with others of larger wattage rating. This step may be especially helpful if the resistors now in the circuit are of half- or quarter-watt rating.

Distortion and Volume Controls

Q. *My power amplifier is behaving in a most peculiar manner. When the volume control is turned up full, the bass response is normal, but as the setting of the control is reduced, the bass falls off sharply. This is not a result of the Fletcher-Munsen effect, for, as I advance the gain of preamplifier at the same time that I decrease that of the power amplifier, the bass still continues to fall off. Enclosed is a schematic of the input circuit of the power amplifier,* (Fig. 1), *perhaps it will help*

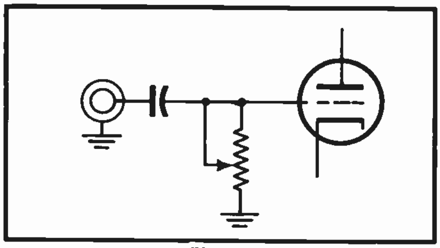

Fig. 1

you to determine what is wrong. James P. Gooley, Oak Lawn, R. I.

A. Notice that in the input circuit of your amplifier the volume control is connected as a rheostat. This circuit can perform the function of reducing the gain of the amplifier, but, as you have seen, the bass will be attenuated in greater amount than the remainder of the signal as the gain is lowered.

To explain why this circuit causes this behavior, let us assume that the reactance of the coupling capacitor is 0.5 megohm at 30 cps, and that the resistance value of the potentiometer when fully open is also 0.5 megohm. At 30 cps, half the voltage developed by the preamplifier will be lost across the coupling capacitor, while the remaining half is available for application to the grid of the input tube of the power amplifier. Let us assume that the gain has been reduced, and the resistance of the control is now 0.25 megohm. Of course the over gain of the amplifier has been reduced 6 db, but our 30-cps tone has been decreased by an even greater amount. The reactance of the capacitor is still 0.5 megohm, but the resistance of the pot has been reduced so now only ⅓ the voltage produced by the preamplifier is available to the power amplifier. This effect will become more and more severe as the resistance of the potentiometer decreases. What is needed is a control circuit which maintains a constant reactance, but still allows the grid to pick off as much signal as is needed, by means of voltage divider action. *Figure*

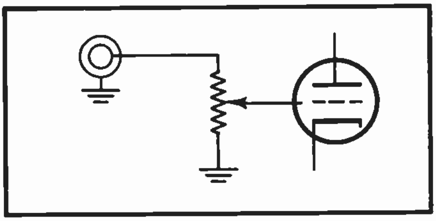

Fig. 2

2 shows how this same 0.5-meg. control can be wired to accomplish this. Note that the coupling capacitor is connected through the full resistance of the pot to ground. The slider has no effect upon the reactance presented to the coupling capacitor, since the grid to which it is attached draws no current and is therefore of infinite resistance. Æ

Volume Controls and Power Amplifiers

Q. I have a power amplifier having an input grid resistor of 1 megohm. I also have two preamplifiers and I would like to connect a volume control to each of their outputs in order to improve signal-to-noise ratio. What value of volume control would you recommend that I use? Neither preamplifier has a cathode follower. Would it be better to locate the volume control at the end of the preamplifier or locate it at the input of the power amplifier? What are the general principles involved in determining the resistance of a volume control? S. W., New York City, N. Y.

A. First, remove the input grid resistor from the power amplifier. Connect a volume control of the same value in its place. (Later in this answer I will outline the means of determining the correct value of control. Check to see whether the control you substitute conforms to this procedure.) Be sure that the grid of the amplifier is connected to the arm of the control. The leads of the input connector on the power amplifier are connected across the full resistance of the control. The method of determining the value of a volume control is the same as that employed when determining the value of a grid-load resistor. It should be at least twice as large as its preceding plate-load resistor. The value of the interstage coupling capacitor is also a factor. Its reactance at the lowest audio frequency you wish to pass should be considerably lower than the grid resistance.

It would be better if the control units employed cathode-follower outputs. This is especially true when the control units are to be located some distance from the amplifier because the length of the interconnecting cable will then be great enough to introduce considerable capacitive reactance which, in turn, will cause some degradation of the high frequencies. This condition does not prevail with cathode-follower circuits largely because their impedance is much lower than the capacitive reactance of the shielded cable, and hence, the shunting effect of this reactance is negligible. Æ

Amplifiers and FM Interference

Q. I live about three city blocks from a fairly strong FM station which does commercial broadcast music via multiplex. This FM signal has been coming through my music system for about two years. By juggling the various a.c. power leads, audio cables, and all concerned, I have been able to keep the amount of signal picked at a minimum.

I have just completed a stereo installation. The FM station is coming in much more strongly, and curiously enough, it is more pronounced in the "left" channel than in the other. In the right channel, I get an overdose of hum, which doesn't worry me. I have been fighting the hum war for years, and I am confident I can lick that. My question is, How can I prevent this annoying interference? Cameron Magnon, Tampa, Fla.

A. The first thing to ascertain is just where the signal is entering the equipment. To do this, first short the pickup leads. If the signal ceases, you have obviously located the source of entry, the pickup and/or associated leads. If the interference still persists, short the grid of the second stage of the preamplifier and so on down the line until you have found the point of entry of the signal.

You will then have to experiment with bypass capacitors. You probably can find one whose value is sufficiently large to shunt out the interfering signal and yet small enough so as not to limit the high-frequency response of the equipment.

If this method fails, place a choke in series with the offending lead. This choke can be made by winding 20 turns of No. 24 enameled wire around a 1-megohm 2-watt resistor. You may have to provide shielding for this filter arrangement. The filter can be rendered even more effective through the use of a second capacitor, arranged in the circuit in such manner that the circuit is a capacitor-input pi-type filter.

Should the interference still continue, you will have to resort to a wavetrap rather than to the filter system just described. This is made of a series-resonant circuit placed across the offending input circuit, and a parallel resonant circuit placed in series with the hot lead. *This unit must be shielded* and the shield returned to a good ground. Capacitors here should be variable, and their values are 10 μμf maximum to 1 μμf with their rotors open. Inductances are wound of No. 14 enameled wire on a ⅜-inch form. Each inductance should contain 3 or 4 turns. The inductance should not be closewound, but rather, should be spaced. After the coils are wound the forms are slipped out, leaving a self-supporting structure. After the wiring of the wavetrap has been completed, set the amplifier to the phono position or whichever position produces the interference, and adjust the tuned circuit for minimun signal.

If no such dip can be found, the coils in the network contain too much or too little inductance. You can determine which of these conditions prevails by compressing the turns. Compression increases inductance, expansion decreases it. If this procedure does not produce a null in the signal, you must then add or subtract turns.

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Power Supply Filtering

Q. The power supply circuit for an amplifier I plan to build calls for an 8-H. choke and an 8-μf capacitor. Because of space limitations, I desire to substitute a resistor for the choke, and increase the size of the capacitor. I would like to know the relative value of RC filtering as opposed to LC filtering. I have noticed that when a choke is used in filter circuits, the capacitors are usually of smaller values than those used with a resistance-type filter. Does the use of RC filtering have any adverse effect, such as increased 120-cps hum or poorer voltage regulation? R. D. Dickson, San Francisco, Cal.

A. The advantage of LC over RC filtering lies in the fact that smaller associated capacitors need be used because, in addition to resistance, the choke possesses inductance which tends to oppose any change in the strength of the current passing through it. The resistor in the RC filter merely serves as a time constant to slow down the rate of change and discharge of its associated capacitors. The choke has some resistance along with its inductance, and can therefore perform this time-constant function to some slight degree. If the capacitor and resistor sizes are made large enough, the ripple content of the power supply can be very nearly that of the LC type. Where small currents are involved, the RC filter circuit works very well indeed. Where the power supply is called upon to furnish current which consists of transient variations, such as in power amplifiers, the RC type filter is definitely *not* recommended, because the filter resistor is so large that, on peak current demand, much voltage is dropped across it. The voltage dropped across the resistor robs the load of the constant voltage needed for proper linear operation.

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Amplifier Power

Q. I've heard a great deal lately about amplifier power, to the effect that more is always better, provided that the power is clean at higher levels. Would this be true for a speaker system of high efficiency, such as a Klipschorn? I'm using a 30-watt amplifier to feed such a system but, if 50 watts of comparable quality would give me better performance, I would like to make the change. J. H. Moore, Tulsa, Oklahoma

A. So long as the power amplifier is of good quality and is operated well below its maximum capabilities and is fed into a speaker of reasonably good efficiency, there is no need to substitute one of higher power output capabilities for that which you are now using. However, if your listening environment is such that this amplifier must be run too close to its capacity, then I should certainly suggest that a more powerful unit be substituted. Be sure that the speaker system is capable of continuous operation at the highest program level to be used, and in fact, some tolerance should be left to account for transient peaks. If your speaker system cannot cope with the demands to be placed upon it, additional speakers should be used which can take up the power and which can provide better sound dispersion, too.

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Amplifier Instability and Remote Lines

Q. This past weekend I was asked to connect a remote speaker system for a friend.

The system consists of an amplifier feeding a 500-ohm line, thence to a 500-ohm speaker and an 8-ohm speaker fed through a matching transformer, each speaker controlled by "T" pads on the speaker side of the transformer. The distribution line was an unknown (but very long) length of No. 18 lamp cord.

At moderately loud volume levels, the system was unstable and would motorboat badly. Reduction of bass would stave off the motorboating some but not much. In connecting an outboard jack to the line I *noticed that holding one lead of the line while standing on wet ground encouraged the instability; touching the other lead had no effect. Also, this effect was not noticeable when standing on a dry board.*

The amplifier was connected to the speaker through a short lead and full gain could be used with no instability.

From the above I assume the instability is due to line capacitance. If so, could this be eliminated without replacing the line, since replacement would be almost impossible? H. S. Newins, Red Bluff, Calif.

A. I agree with you that the instability is caused by an alteration of the feedback characteristics of the amplifier resulting from the long line. It will be hard to say whether this trouble is the result of capacitive effects or inductive effects because a long line will contain significant amounts of both.

Before adding reactances and capacitances in an attempt to tune this difficulty, ground the amplifier to a good ground, and ground the common side of the far end of the 500-ohm line and one side of each speaker voice coil. Sometimes this kind of grounding will shunt out this kind of instability. If it works, it will save you much trial and error fiddling with inductances and capacitances which will otherwise be your fate.

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Connecting Headphones to Amplifiers

Q. I would like to know how to connect a pair of 600-ohm headphones to my Brook amplifier. John Sabritt, Philadelphia, Pa.

A. The Brook amplifier is a special case in that it includes a 500-ohm winding in its design. Furthermore, this amplifier is fitted with an octal plug, rather than the conventional tie strip used to connect loudspeakers and other devices to most other amplifiers. Pins 7 and 8 on this output plug represent the 500-ohm winding, and your phones may be connected directly to these pins.

However, there is another factor which should be taken into account. Earphones are placed very close to the eardrums, thereby creating a very efficient coupling network. This means that only a small amount of power is necessary for good listening level in the headphones. In fact, the power required for adequate listening level is approximately 9 milliwatts, or 9/1000 of one watt. This means that the volume control need be turned but slightly to produce such a low power level in its output circuit. This small degree of volume control rotation will make it difficult to operate your amplifier smoothly.

This difficulty may be overcome through the use of a pad. It may be constructed as follows: Connect a 500-ohm resistor across pins 7 and 8 of the output plug on your amplifier, the 500-ohm tap. Use a 5-watt wire wound resistor. Connect a 4700-ohm, 1-watt resistor to pin 7. Connect a similar resistor to pin 8. Connect the free ends of these resistors to the headphone terminals. By this means you will have introduced a loss into the system, a voltage division of 20 to 1. Naturally, considerably more power will be needed to drive the headphones with this pad in the circuit.

The pad has another advantage. Because of the high sensitivity of the headphones, any residual hiss level in the amplifier will be present as an annoying background to the program material being listened to. The pad will attenuate this background noise to a level such that it will be barely noticeable.

Other amplifiers are not provided with a 500-ohm winding, but these also may be connected to 600-ohm phones. The amplifier again should be properly loaded with a resistor connected across the output tap selected. If, for example, the 8-ohm tap is to be used, it should be of perhaps 5 watts capacity. A similar voltage-dividing network should be used between the 8-ohm resistor and the headphones. If you wish to experiment in order to obtain the best signal to noise ratio, make the resistances variable. Once you have selected the desired settings, measure the resistors and substitute fixed values.

The headphones may also be connected directly to the output of your preamplifier. 600-ohm headphones will unfortunately load down most cathode-follower circuits, and this loading will cause a degradation of low-frequency response, which is already badly degraded because of the nature of most of the headphones employed. High-impedance phones, especially crystals, may be connected directly to the output of the preamplifier without such degradation.

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Component Life

Q. *I have a Heath Kit W5M amplifier, and WAP2 preamplifier. When these components were new, the noise level was inaudible unless one searched for it. Now I am beginning to notice a little high-frequency noise. It has not come to an irritating level as yet. It is a case of where before there was none, now there is some. I am assuming that this noise is caused by the gradual breakdown of resistors or capacitors, as I have replaced all the tubes* *and the noise remains. How long should one expect carbon resistors, paper capacitors, electrolytic capacitors, ceramic capacitors, etc., to perform their functions satisfactorily? How long should one expect tubes such as those used in the equipment described, to last under normal conditions? Clyde A. McGoldrick, APO, San Francisco, Cal.*

A. The usable life of any component depends its initial quality and upon the conditions to which it is subjected. These might include temperature, moisture, and the degree of closeness to its maximum rated capacity at which it is operated. Resistors operated well within their ratings may last longer than ten years. Waxed paper capacitors may be expected to last about five years, although in many cases their lives are much shorter. With these components, temperature and moisture are most important. I should expect molded paper capacitors to last close to ten years when conservatively operated. Mica and ceramic capacitors may well last more than ten years. The life expectancy of electrolytic capacitors varies greatly. High-grade units may last five to ten years, and sometimes longer, while some units last barely a year.

Tubes are in a class by themselves, and hence will be given special treatment. When used for periods of perhaps an hour or two a day, they may be expected to last three to four years. However, they may last for much longer than that. I have had tubes in my FM tuner for ten years, and still others in my communcations receiver for 12 years, and they are still performing satisfactorily. Rectifier tubes are usually the first to burn out.

Tube life may be greatly lengthened if the proper measures are taken before the tubes are placed in service: Place the tube in a convenient and appropriate socket. Many of you can salvage it from an old chassis or from the junk box. Preferably, the tube should be in a vertical position. Connect the heater terminals to a source of power whose voltage can be varied. Over a period of a week gradually advance the heater voltage from zero to normal. Do not apply plate voltage. Allow the heaters to remain at their normal voltage for another week. The longer the tube has stood on the shelf, the greater will be the need for this aging process. Tube manufacturers have neither time nor facilities for this kind of aging. They connect the heaters of the tube, for a short time, to a source of voltage higher than that recommended for normal use. This process, because of its extremely brief duration, is known as flash aging. Its purpose is to cause the movement of more electrons to the surface of the cathode, thereby improving the emission capabilities of the tube.

Another aid to tube life is never to turn off the equipment. Remember, though, that the money saved on tube replacements will be much less than the electric bill run up by leaving the equipment running.

The data presented here represent information gained while servicing equipment.

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Tuner Problems

Q. 1. The AM section of my tuner produces more hum and background noise than the FM. I have heard other tuners in which the AM section is quiet. 2. When I connect the FM section of the tuner to my double conical TV antenna, I receive the same station at many places on the dial. What is causing these two conditions and what can I do about it? Robert McDonald, Oakland, Calif.

A. 1. The hum which is present in the AM section of your tuner can be caused by several things: perhaps the AM section is not well filtered, leading to the supposition that perhaps one of the filter capacitors has become defective. There may be a leak between the heater and cathode of one of the tubes. It may be generated as a result of poor grounding or oxidized house wiring. If this latter is the case, I don't believe there is much you can do, especially if you are an apartment dweller as I am. There is also the possibility that the hum is caused by something on the line to which the tuner is connected. I have an AC-DC dictating machine which, when turned on, introduces hum into every AM receiver in the house. Hum arising from these last two sources can sometimes be minimized by the use of an outside antenna. It need not be elaborate. Make it about 20 feet long and keep it well insulated from surrounding objects. The lead-in wire from the antenna should be of coaxial cable, so that the lead-in cannot pick up any interference. Naturally, the shield of the cable should be returned to a good ground, as should your tuner chassis.

The background noise of which you spoke may have its roots from many places. Manmade interference, such as that produced by vacuum cleaners and fluorescent lights, probably heads the list of possible candidates. (Next month I shall discuss a special type of background interference, that of the direct radiation of harmonics of the horizontal oscillator.) Background noise can be generated within the tuner itself as a result of defective tubes, coils, r.f. and i.f. transformers, resistors, capacitors, and the like. The stage which, in my experience, most often causes this trouble is the mixer stage.

2. The reason you receive one station at several places on the dial is that the front end of your tuner is misaligned, poorly designed, or overloaded by excessively high input signal levels. I suspect the latter possibility because your antenna is very efficient. A straight dipole or folded dipole will probably give more than adequate results. Keep the dipole out-of-doors if possible because, when it is indoors, passersby may cause the desired signal to be reflected away from it, leading to fading and fluttering.

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Delayed AVC

Q. What is delayed AVC and what is its purpose? George Lystad, Lake City, Ark.

A. The purpose of AVC is to keep the audio output of the receiver relatively constant, regardless of the strength of the received signal. This is accomplished by rectifying the signal voltage and connecting it in such a manner as to make the grids of the r.f. and i.f. stages more and more negative as signal strength increases. Thus, the receiver is more sensitive to weak signals than it is to strong ones. No matter how weak the signal, some voltage is fed back, thereby reducing the sensitivity of the receiver. Obviously, when reception of weak signals is to be accomplished, the receiver must operate at maximum sensitivity. This cannot be done when the AVC is operating, as previously noted. Therefore, means must be provided to make the AVC operate only when signals reach at least a moderate strength. This can be done with a manual switch connected in the AVC bus which removes it from the tube circuits. There is also an automatic means for accomplishing this. Two diodes are needed for this circuit. One is connected in the conventional manner, and is used to demodulate the signal. The other is used only to develop AVC. The plate of this diode is biased negative with respect to its cathode by the desired amount. Weak signals will not develop sufficient voltage to overcome this bias but, as signal strength increases, the diode will conduct and form AVC in the usual manner. Since the AVC is not operative until the signal reaches a predetermined strength, this system is known as delayed AVC (DAVC). The use of two diodes is mandatory here, since if only one were used, it would fail to conduct at all on weak signals, and even when it did start to conduct, less than half of the cycle would be reproduced, leading to distortion.

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Dynamics, AVC, and Broadcasting

Q. I believe I understand the principle of the action of AVC circuits in radio receivers. However, one thing puzzles me. Wouldn't this action operate on the signal itself and compress the dynamic range of the program material. I have never yet heard a radio broadcast that sounds like a phonograph; there is always some sort of degradation which transmitter and receiver distortion does not always explain. William Devine, Detroit, Mich.

A. When AVC is used with AM receivers, the filter time constants are so chosen that the rapid modulation peaks are not smoothed out, but slow variations in carrier strength are. When the dynamics increase, the average strength of the carrier is maintained, whereas the peaks and troughs increase in size. If the time constants are chosen to be 100 milliseconds, frequencies as low as 20 cps can be transmitted without the AVC action smoothing them out unduly. As the time constant is shortened, more and more of the lows will be smoothed out, leading not to a reduction of the dynamic range, but to an erasure of the lows from the audio output.

When the volume varies during FM broadcasting, the sizes of the peaks and troughs remain constant. Volume increases manifest themselves as increases in frequency deviation from the center frequency. It is these frequency deviations which are detected, rather than any change in carrier amplitude. It should be obvious now that the clamping effect of the AVC and limiters can in no way impair the dynamic range of the program material.

Compression of dynamics probably comes from the broadcaster's desire to maintain as high a signal-to-noise ratio as possible.

Tuner Sensitivity and Quieting

Q. *Can you tell me whether it is possible to have an FM receiver which will completely suppress ignition noise interference even when the latter appreciably exceeds the desired signal? If not, what type of circuit most nearly approaches this ideal? Can you explain exactly how the sensitivity in mv for 20 db quieting is obtained for FM tuners? Judging by the advertisements in* AUDIO *and other magazines, there are two or more methods in use. What does absolute sensitivity mean? What is the theoretical limit of sensitivity of an FM tuner? B. H. Murdoch, Belfast, Northern Ireland*

A. The only answer I can give with regard to a tuner having perfect suppression, is that there is no such thing as perfection. Any limiter can be upset when overloaded with noise signal. The amount of signal needed to accomplish this will depend upon both the strength of the desired signal and upon the design of the limiting circuit. Probably the closest approach to this ideal is the discriminator circuit, preceded by two limiters. Following very closely behind this is the ratio detector preceded by a single limiter and containing stiff AGC. As a matter of fact, the discriminator circuit should also employ some AGC, especially at the front end. This is because a tuner must handle signals of greatly varying strength. In the New York area, for example, it is possible to receive signals as strong as 0.5 volt (yes, ½ volt) at the antenna terminals. At the other extreme, the tuner must accept signals as small as 2 microvolts and still manage to quiet satisfactorily. That's asking a lot of a front end, and it is the reason that AGC is very much needed. Notice that with the discriminator circuit, two limiters are needed, whereas with the ratio detector, only one is needed. This comes about because the ratio detector circuit has inherent limiting properties. Further limiting would only reduce i.f. gain, and this would serve no useful purpose.

Now, let's go into the problem of sensitivity and quieting. There are several methods for measuring these two quantities. When these are applied properly and interpreted correctly, they mean much the same thing. The standard employed by the Institute of Radio Engineers may be summarized as follows: What voltage, when fed into a 300-ohm input, will give 30 db quieting, when 22.5 kc deviation is applied? Notice that there are two other terms which must be taken into account besides input voltage and the number of db of quieting. These are the input impedance and the amount of deviation, or percentage of modulation. If we make our measurements at an impedance of 72 ohms and feed the same power to the input, the voltage appearing at the antenna terminals will be half that which would be obtained at the 300-ohm impedance. This means that if our tuner requires four microvolts for 30 db of quieting at 22.5 kc deviation and at an impedance of 300 ohms it will need only two microvolts for the same degree of quieting when an impedance of 72 ohms is employed. This sounds like an improvement, but a 72-ohm antenna system gives us only half the signal voltage provided by a 300-ohm antenna system. Naturally it is assumed that both antennas are of equal efficiency; and that they are in identical locations. In other words, the two measurements are, for all purposes, identical.

Next, we come to the matter of deviation. The I. R. E. used 22.5 kc because it corresponds to 30 per cent modulation. This, in turn, is roughly equal to average program level whose peaks are 10 db higher, equaling 100 per cent modulation. This figure was selected because people listen to average program level most of the time, rather than to peak levels. The Institute reasoned, therefore, that noise impairs average level more than it does peak levels because the average signal strength is weaker than peaks. Other methods, however, make use of a deviation of 100 per cent, 75 kc, as the basis for their quieting measurements. This gives us an apparent improvement of slightly more than 3:1. Our tuner which required four microvolts for 30 db of quieting will now require only 1.333 microvolts and actually slightly less, for 30 db of quieting. If we use an antenna system and input circuit designed for 72 ohms impedance, we will have a tuner requiring only 0.666 microvolt for 30 db of quieting. That sounds like a pretty good tuner, but it's no better than our original model, or should I say, "no better than our original specifications," since we have really done nothing at all to the tuner. Actually, it's all in how you interpret the figures.

We can go even further in our direction of smaller and smaller input voltages for good quieting. All we need do is to assume that good limiting can be had with 20 db suppression, rather than 30 db. We need only ⅓ as much signal to obtain this degree of suppression, and our figures are growing small indeed, but then, so is our suppression. There are other factors which affect suppression.

There are other factors, however, which caused the standards committee of the Institute of High Fidelity Manufacturers, Inc., to propose a new set of standards. They note that as the signal strength decreases, not only does the noise increase but so does the distortion. This increased distortion is largely caused by a narrowing of the i.f. band-pass. Therefore, the standards committee of the Institute of High Fidelity Manufacturers conceived the idea of a total usable signal measurement. The method for making this measurement can be summed up as follows: What signal voltage, at an impedance of 300 ohms, and at a 75 kc deviation, will be required to cause a signal at the output of the tuner which shall consist of 3 per cent total noise and distortion? The method described takes both these factors into account, and this method is quite valid. However, it certainly is going to add much confusion to already troubled waters.

Lastly, you wanted to know what is meant by absolute sensitivity. It is approximately 0.71 microvolt for 20 db of quieting, at a deviation of 22.5 kc. The reason that no greater sensitivity is possible is that the input circuit will contain noise of its own which will be 20 db below this value. It should be stated that this measurement is based upon an input impedance of 300 ohms. Any impedance generates its characteristic amount of noise, and there is no way we can prevent this, unless this impedance were placed at a temperature of absolute zero, which would of course cure the random noise generated across the impedance, but so placing the impedance would pose grave problems for the tuner manufacturer.

When making any measurement where impedance is a factor, it is important that the input to the tuner be truly matched if valid results are to be obtained. Most signal generators have outputs of 50 or 72 ohms. Equal values of resistance should be placed in each leg of the generator, and the total should equal the impedance of the tuner input. Only balanced generators should be used, since an unbalanced unit will introduce standing waves which will affect the validity of the measurement. If an unbalanced generator were used, however, all the padding resistance would have to be placed in the hot side of the line.

Interference from TV receivers

Q. *When I tune my receiver through the AM band, I find it covered with a series of whistles whose pitch varies according to the station to which I am listening. Since this phenomenon occurs with many other receivers in the apartment house, I don't think it is because of any possible misadjustment of my whistle filter. The only time the whistles do not appear is early in the morning. When my TV set or someone else's, is turned on, the whistle reappears. What is causing this, and what can I do about it? Woodrow C. Doebler, Pottstown, Pa.*

A. This series of whistles is the result of beats between broadcast stations and harmonics of the horizontal oscillator of the offending TV receiver. Since the space between harmonics is small (the frequency of the horizontal oscillator being 15,750 cps), they can beat with any station.

There are two ways by which the harmonics can enter your receiver. The first of these is by direct radiation from the TV set, and the harmonics of the oscillator are picked directly by the receiver's antenna. This effect can sometimes be minimized by placing your antenna high enough so that it will be out of the radiation field. (Some receivers have been known to radiate several hundred feet, in which case you cannot erect an antenna high enough.) This can also result in the strength of the desired broadcast stations' being great enough to override the whistles. The antenna should be fed by coaxial cable, so that no TV set radiation will be picked up by the lead-in.

The second and less unlikely means of entry of these undesired signals is through the power lines. This may be overcome by inserting an r.f. choke in each side of the line feeding your tuner, and bypassing all choke leads to ground with good-quality mica capacitors. Æ

Radio Volume Control Considerations

Q. *When the volume setting of my radio receiver is changed rapidly, I notice a considerable delay between the time I make the volume change and the time when the change becomes audible. What could cause this? Madelain Gold, Chicago, Ill.*

A. *Figure* 1 shows a typical volume control circuit of an AM radio receiver. The potentiometer, R_1, is a portion of the diode load and is therefore carrying a steady

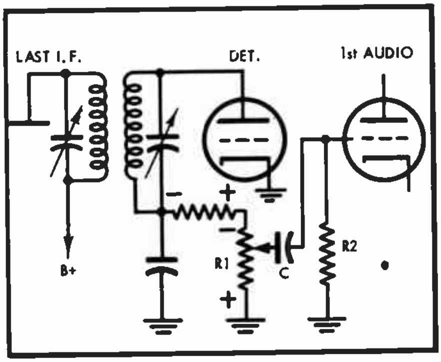

Fig. 1

direct current. As the control is advanced, a greater and greater charge is applied to coupling capacitor C, which charge is transferred to the grid of the first audio stage. If the grid resistor for this stage, R_2, were infinite, this charge could never leak off, and the tube would be cut off. As R_2 is made smaller, the charge can leak off more and more easily. The size of the coupling capacitor also determines the time needed to discharge through R_2. If the capacitor is large it will have to take more time to discharge, since it took a greater charge initially. If the discharge rate is low there will be a negative voltage applied to the first audio grid every time the volume control is turned up, which may completely cut off the stage until balance is once again reached. Some may say that the capacitor is intended to block d.c. from ever getting onto that, which is true. Remember, however, that, so long as the capacitor is being charged, it does not exist. The process is something like having a resistor substituted for the capacitor. When the d.c. is first applied, the resistance is zero, gradually rising to infinity, as the charge builds up to its maximum. In practice infinity is never reached, since there is always a certain amount of leakage present within the capacitor itself.

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Matrixing

Q. I have heard much lately about matrixing. Just what is it? G. Best, Long Island City, N. Y.

A. Matrixing is a system whereby two signals are combined to form their sum and their difference; later they are reconstituted into their original components. This technique is employed in the Crosby multiplex system. The sum of two stereo channels is fed into an FM transmitter in the normal manner. (Those who don't have multiplex adapters may, by this means, receive the monophonic broadcast.) The difference signal is impressed upon a subcarrier, which is transmitted on the main FM carrier. This subcarrier is undetectable with unmodified FM receivers, but this difference information can be recovered by adding a multiplex adapter to the tuner. Some circuit modification will be needed on tuners which were not provided with a multiplex jack. One of the things which is included in the adapter is a matrixing circuit, which serves to recover the difference signal and recombine it with the sum signal in such a way that the original stereo information is recovered and can be fed to the stereo preamplifier in the usual manner.

Another use found for this technique is the Columbia Record stereo system, which matrixes the signal and records the sum signal laterally and the difference signal vertically, with compression on the difference, or vertical channel.

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Multiplexing

Q. I recently bought an FM tuner on which there is an output marked "multiplex." Can you tell me to what purpose this may be put? R. P. Burns, Havertown, Pennsylvania.

A. Multiplexing is a system by which two or more programs may be transmitted simultaneously by the same FM transmitter. In one of the several methods now in use, one of the two channels is transmitted in the normal manner, while the intelligence of the other channel occupies a bandwidth of 15 kc. in the supersonic range. This channel is separated from the normal channel at the detector by means of a sharp cutoff high-pass filter. After this separation process, it is heterodyned with another oscillator in such a way that the beats fall into their pitch relationships in the audio spectrum, and can therefore be heard. This is just one of several methods which are in use, or which are under study, for the simultaneous transmission of two or more programs. Until one method is adopted as the standard to be used by all interested parties, little, if any, equipment will be available for detecting the second channel.

The multiplex output is wired to the detector circuit in such manner that the de-emphasis network is bypassed. If your amplifier is poor in high-frequency response, you may connect it to the multiplex output. FM reception will be normal except for an exaggerated emphasis of the highs, which can be helpful in systems where response in that region of the spectrum is poor. Let me make it perfectly clear that connecting your tuner in this manner will not allow you to hear the multiplexed channel. You will hear the FM programs to which you are accustomed, but with an overabundance of highs.

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Remote Cartridge Circuit

Q. My layout is such that I keep by Rondine hysteresis motor and Pickering cartridge in another room, quite removed from the amplifier. As the leads are fairly long, I fear considerable loss of highs in the shielded cable. Could you suggest a simple circuit using, perhaps, a 12AY7, with one stage of flat amplification and a cathode follower stage using d.c. on the heaters? Frank Gittelson, Lynbrook, N. Y.

A. *Figure* 1 should solve your problem.

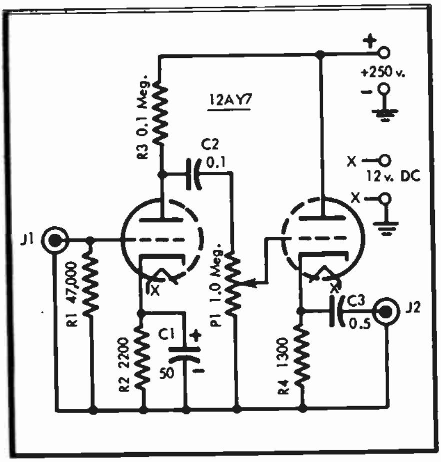

Figure 1

The circuit employs a 12AY7, chosen because of its low noise content. It is used as a voltage amplifier and cathode follower output. Thus, the pickup is presented with its proper impedance and the output impedance is sufficiently low to enable the use of lines up to 50 ft. in length or even more, without serious high frequency degradation. Because of the possibility of overloading your preamplifier, a volume control was incorporated in the cathode follower stage to limit the signal feeding the preamplifier to a level somewhere below the fold-up point. The circuit is designed to operate with approximately 250 volts on the plates. Since the plate and filament supplies for this circuit are entirely conventional, they are not shown. This circuit may also be used to feed other high-impedance devices into the preamplifier. Care must be taken not to overload the 12AY7 and means must be provided for adjusting the value of the input resistor, should the output impedance of the device used be in excess of 47,000 ohms. Æ

Matching Transformer Impedances

Q. The impedance of my phonograph cartridge is very low, resulting in output voltage insufficient to drive my preamplifier. I therefore resorted to the use of a matching transformer. However, I noticed that when the secondary is terminated in a resistance of the proper value, less output is obtained than is had when the transformer is terminated in a resistance several times larger than that of its nominal secondary impedance. What is the significance of these findings? S. Kalmer, New York City

A. We match transformer impedances mainly with the idea of transferring as much power from one circuit to another as possible. This can be done only when the transformer is terminated in a resistance equal to the nominal impedance of the transformer. However, a vacuum tube is not a power sensitive device but is, rather, voltage actuated. A transformer will, when terminated in a resistance higher than its proper terminating resistor deliver a higher voltage into that resistance than it would into its correct value. Although this happens, maximum power is no longer transferred. While maximum voltage will occur when the transformer is terminated in a resistance at least five times its nominal rating, the low-frequency response is optimum only when the transformer is properly terminated, since the low frequency response is determined in part by the current flowing in the windings of the transformer. Exactly what will happen to the response varies from unit to unit and so, no figure can be given. If the additional gain is not required, terminate the transformer in its proper load. If it is needed, adjustment of the bass controls should be carried out by means of trial and error.

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Stereophonic or Monophonic Sound

Q. I have many monophonic discs and tapes. Now, with the advent of stereo, is my monophonic collection worthless? Arthur Darrow, Albany, N. Y.

A. I have met many people who lament the fact that stereo has made their fine record collections obsolete and miserable. While it is unquestionably true that the addition of spaciousness to music adds much to our emotional reactions to it, it does not and should not mean that we can no longer enjoy our otherwise fine discs. When you stop and think about it many people began collecting records in the 20's and before, and the sound on those early discs was poor indeed compared to those of today. Those people have not discarded them. I guess this is partly related to sentiment and is the symbol of a past which many of them considered to be better than the present. Probably, though, in the vast majority of instances, people hold onto these discs simply because of the artistry of those appearing on them. Stereo, wonderful though it is, cannot give us those oldtimers who have flashed across the concert and popular stages. This is not simply true of the 20's. It holds even for comparatively recent monophonic releases. What about those releases of old 78's or the immortal performances of Toscanini? True, stereo could have enhanced all of these performances, but they are still fine, valid ones, even without stereo. Of course, a record need not be world-acclaimed in order for it to be enjoyable to you. The main thing is that you liked it when you bought it and probably did right up until the time you heard your first stereo broadcast. Listen to that monophonic disc or tape again and you will probably enjoy it as much as always. I have such discs and tapes in my own collection, and I have stereo as well.

If possible, you should have equipment capable of playing both types of material. After all, there will be new music and new performances of old music. All of this will be recorded for our enjoyment. It will be captured in stereophonic sound.

You should not feel that the stereo system is just a flash in the pan, since material is coming at us with extreme rapidity, and this material and the equipment with which to play it, is being sold at a tremendous rate. It is hard to believe, but in the short time that the stereo disc and tapes have been with us, over 1,000 titles have been released.

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Distortion in Monophonic Discs

Q. I have been a record collector for a good many years, and, as you can imagine, have a great number of monophonic discs, in addition to a few of the new stereophonic discs. Because this stereo system is compatible, I use my stereo cartridge to play both types of discs. However, many of the monophonic discs are heard with a surprising amount of distortion as contrasted to their sound when a conventional, monophonic cartridge is used. What is causing this distortion and what can I do about it? James Blake, San Francisco, Cal.

A. There are several possible causes for the distortion of which you speak. One is that you are using insufficient tracking force. Some stereo cartridges are constructed in such a manner that they must track at or slightly above the prescribed force in order for the stylus to be properly oriented with respect to magnetic pole pieces. Failure to observe this precaution means that the cartridge will be too compliant, and the stylus will ride out of the grooves, rather than tracing them faithfully.

The second cause of distortion is a function of the monophonic discs themselves. Many of them are greatly overcut, especially at the high end of the audio spectrum. This will result in the inability of the stylus to stay in the groove properly. It tends to ride up out of the groove, regardless of tracking force. Any time that the stylus rides upward some vertical signal will be produced. Because of the nature of the system, this vertical component will transmit it to the loudspeaker. Most monophonic cartridges were specially designed to eliminate as much vertical output as possible, and for this reason, little, if any, of this form of distortion was detectable with your monophonic cartridge.

This trouble can be minimized by connecting the two sections of your cartridge in parallel, phased in such a way that the vertical output is cancelled. Most instruction sheets supplied with these stereo cartridges show a wiring configuration which will bring about this end. Naturally, if you wire this cartridge in this manner, you cannot achieve the stereophonic effect. What you will have to do is to wire a switch and mount it in some convenient place. This will enable you to switch from stereophonic sound to monophonic sound. If there is sufficient interest, such a switching circuit will appear in a future column.

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Turntable Speed

Q. I have heard much discussion about how the size of the intermediate idler of a phonograph turntable will affect the turntable's speed. Is this so? I have seen tables which slow down as the idler is pressed more firmly against the motor shaft, thereby making the idler smaller. Mark W. Tennberg, Albany, New York

A. The size of the idler has no effect upon turntable speed. The reason a table slows down when more pressure is created between the idler and the motor shaft is not that of the idler's possible reduction in size, but simply because the motor is loaded more heavily under those conditions, with this loading, in turn, causing the motor shaft to turn more slowly.

In order that I may give reasons for my position, picture a motor shaft whose diameter is one inch. This, in turn, drives an idler whose diameter is one inch, which, in turn, drives a turntable whose rim at the area of contact with the idler is also one inch. Because the ratios of the motor shaft, idler, and turntable are 1: 1: 1, all wheels will rotate at the same speed. Assume now, that the idler has worn down to a half inch in diameter. The ratios are now 1: ½: 1. This means that, for every revolution of the motor shaft, the idler to which it is friction coupled must revolve twice. It would seem that the turntable must rotate faster under this condition. This is not true. The turntable is twice the diameter of the idler, which means that the idler must revolve twice in order that the table may complete one revolution. In order for the table to make one revolution the motor shaft makes one revolution, the idler now makes two revolutions, while the table makes one.

Should it still be hard to picture the foregoing, think of the parts in terms of their linear distances, which are their circumferences. Assuming no slippage, the number of linear inches traveled by a given point on the motor shaft must equal the number of linear inches traveled by points on each other part of the chain. Work out the ratios in terms of linear inches which must be covered by each part, and you will see that the turntable has covered the same number of inches, regardless of idler speed.

Do not, as so many people try to do, apply this explanation to belt-drive systems in which the motor drives a pulley which is belt-driven to another pulley, on the same shaft with which is located a third pulley belt-driven to a fourth, and so on. Each belt must be considered a closed, separate loop. In order for the above discussion to be applied to belt drives, it would be necessary for the intermediate pulleys to be common to two belts. This also applies to stepped idlers—those that have two or more different diameters.

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Ghost and Echo

Q. What are ghost *and* echo*? Jay Sharpe, Oakland, Cal.*

A. These are annoying disturbances which can occur as over-recorded discs are played back. When an instantaneous lacquer disc is cut, a spiral is cut into the surface. As signal is fed into the cutting head, the cutting needle moves laterally in accordance with the frequency and amplitude of the program material. Because of this motion, the spiral generated is no longer uniform, so that at certain times the grooves are closer to each other that at other times. Lacquer is fragile, and so, if the sideward motion is made too great (over-recording), the wall of the preceding groove is broken through, or at least deformed slightly, and the playback stylus will not track properly. If the wall is not actually cut into, it can become distorted, with this distortion taking the form of the modulation envelope being impressed. In this manner, the groove immediately preceding the one in which over-recording occurred will have both the impression of the signal originally intended and that of the signal intended for the groove following. This latter impression is not as loud as the desired signal, but is nevertheless quite audible. These faint tracings are known as ghost.

Echoes are created as the cutting needle swings in the direction of what will be the succeeding groove. The needle hits the land on this half-cycle so violently that internal pressure is built up, whose amount varies in accordance with the modulation of the groove being recorded. Most of the time, this pressure is removed when the following groove is cut, but in some instances it remains and distorts the newly cut groove wall to the shape of its predecessor. Playing back such a groove reveals a faint trace of the material just heard. This faint after-sound is known as *echo;* it is necessarily far more rare than ghost, because of the tremendous force needed to distort uncut acetate or lacquer.

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Tape Recorder Boss Response

Q. I have made many tests with tape recorders, and I have noticed that none of those I tried have much bass below 30–40 cps. Is this because the tape is unable to accept frequencies below these values? James C. Bryson, Redwood City, California

A. The tape can accept frequencies as low as necessary, provided we take proper care in designing our associated recording and reproducing amplifiers. Characteristically, tape rolls off at the rate of 6 db per octave of frequency decrease. To correct for this we must design our reproducing equipment so that it possesses a boost of 6 db per octave of frequency decrease. This will balance the rolloff and we will have a flat response. Most amplifiers, however, are designed to boost 6 db per octave but only down to a certain point, say 50 cps, after which point the amplifier behaves as though it were flat, providing no further boost. This means that from there on, the bass rolls off at 6 db per octave.

There is no reason why the tape cannot accept tones lower than 30 cps. To illustrate this, let us assume a tape recorder running at a speed of 30 ips. We feed a test signal into the recorder at 100 cps. When the tape is played back at the same speed, we obviously have a tape whose program content is 100 cps. If the tape were slowed down to 15 ips, a 50-cps tone would be noted. If again the tape is played back at 7.5 ips, a 25-cps signal will be picked up, but you'll have to turn up your gain to get it, because your playback amplifier is no longer boosting complementarily to the tape's rolloff characteristic. 3.75 ips will give us a 12.5 cps tone, while a tape speed of 1.875 ips yields a tone whose frequency is 6.25 cps. This speed reduction versus frequency could be carried further till a speed of 0.0000001 *inch per minute* is reached, but I think it should be clear that we can, if we want to, arrive at any low-frequency limit we wish, so long as we design an amplifier with sufficient correction.

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Tape Direction

Q. *In* AUDIOCLINIC *in September, 1957, discussing conversion to stereo, you say that with the shiny side of the tape facing the viewer, the upper half is intended for the left speaker. Which way is the tape running: from left to right or from right to left? Paul M. Gerhard, Beverly, Mass.*

A. The upper channel always feeds the left speaker as the listener faces the speakers in listening position, and it is assumed that the tape is travelling from left to right.

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Microphone Phasing

Q. *I own a microphone mixer capable of handling the outputs of two microphones. The mixer is quite stable in that the channels do not interact with each other. I determined this by connecting a microphone to one channel and, while leaving the other input open, rotating the gain control to maximum while talking into the microphone. Rotating the control caused no change in the output of the driven channel. I checked the other channel with the same result. Yet, when I connect microphones into both channels and talk into both of them, increasing the gain of either channel causes the output to decrease. What would cause this undesirable effect? Raymond E. Leonard, Poughkeepsie, N. Y.*

A. In all probability, the microphones are out of phase with each other. If this is so, output from both channels will combine in the output of the mixer in such a manner as to cancel each other partially. This condition is easily remedied by reversing the leads to *one* of the microphones. In the case of microphones which make use of two-conductor shielded cable, interchange the connections of the conductors, leaving the shield connected to its grounding point. If the microphones have selectable directional characteristics and are close enough to be in the same sound field, they should be set to have the same pickup pattern. When bidirectional microphones are employed, be sure that all action takes place on the same side of both microphones.

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Spurious Response in Recorders

Q. *When a high frequency sine wave is recorded and played back through the recorder, a spurious response is heard which increases in strength in proportion to the fundamental, when the recording signal strength is increased. The frequency at which this phenomenon begins depends upon the quality of the recorder, and I have heard it start at 10,000 to 15,000 cps. I'm rather doubtful that this is a heterodyne effect occurring between the bias oscillator and the input signal, since I heard a response of estimated 5000 cps in addition to the 15,000-cps sine wave tone fed into the machine's input, when recorded on an Ampex 601, which has a bias frequency of 100 kc. If this were a heterodyning effect it would require approximately a sixth harmonic of the 15,000-cps input signal to combine with the bias frequency to give even a 10,000-cps spurious response, which seems unlikely. What is your explanation of this phenomenon? Burton W. Byler, Oreland, Pa.*

A. The 5000-cps tone you hear when a 15,000-cps signal is fed into the tape recorder is caused by the combining of a harmonic of this tone with the bias frequency. Despite the fact that the oscillation fed into the machine is a pure sine wave, harmonics may nevertheless be generated. The greater the level of the recording signal, the greater the harmonic distortion generated within the recording amplifier, and at the head where the signal is mixed with the bias. The bias frequency, furthermore, is only approximate. It is not at all impossible for it to be either 95 or 110 kc rather than the specified 100 kc. These frequencies can heterodyne with either the sixth or seventh harmonic, with the resultant 5 kc tone you have observed. It is also possible for the audio oscillator to be inexact as to frequency. A little figuring will show that an error of slightly less than a kilocycle can produce sufficient error at the sixth harmonic to cause the 5-kc tone, assuming that the bias frequency is exact.

Should you wish to determine precisely the frequency of the bias oscillator, couple some of its signal into the antenna terminals of a standard broadcast receiver, being careful not to introduce d.c. into the antenna coil. You will observe that a beat will occur at several places on the dial. This beat is caused by a harmonic of the oscillator combining with one of the stations. Where there is no station to beat against, the bias oscillator signal will appear as an unmodulated carrier. The frequency separating the appearances of successive signals from the oscillator is determined by the frequency separation of two successive harmonics, which, in turn, is equal to the fundamental frequency of the bias oscillator. The more accurately calibrated your receiver dial is, the more accurately the frequency of the bias oscillator can be determined. Of course, when two successive harmonics of the bias signal beat with two broadcast stations each of whose frequency is known and when each of the beat frequencies resulting from said combination is known, the exact bias frequency can be obtained without accurate receiver dial readings.

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Tape Hiss

Q. *What is the exact cause of tape hiss? This objectionable noise is present on many recorded tapes and even on some professionally-recorded original tapes. Burton W. Byler, Oreland, Pa.*

A. There are two basic causes: 1) The recording amplifier may contain tube and resistor noise whose volume is sufficient to cause it to be recorded on the tape in the form of hiss. 2) Magnetic tape may be considered to be composed of an almost infinite number of minute magnets. The number of these whose polarity and strength is used determines the nature of the recorded material. Some of the magnets are not used; their poles are aligned in a helter skelter manner. Because this alignment is non-uniform, it is obvious that small random groups of molecules will be aligned in the same direction, thereby combining to form larger magnets whose strengths are sufficient to cause voltages to be developed as the tape is being reproduced. These voltages are heard as hiss.

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Crossover Distortion

Q. *What is crossover distortion? William Adsen, Tampa, Fla.*

A. Crossover distortion is not, as might be supposed, an alteration of sound created within a network used to divide the frequency spectrum for use with two or three-way speaker systems. To make clear what crossover distortion is, we must reexamine some of the basic ideas concerning class-A and class-B amplifiers.

The tubes in a class-A amplifier operate approximately midway between the point where grid current flows and cutoff, where plate current flow ceases. This is a static condition which changes when a signal is applied to the grid circuit of the stage. At this time, the current in each tube no longer is equal to the current in the other tube of the push-pull pair. During the first half cycle, the current in tube 1 increases, while that of tube 2 decreases. During the opposite half cycle, the roles of the tubes reverse. The signal magnitude is such that the tubes are never driven into grid current, nor run down to plate current cutoff.

The class-B amplifier poses an entirely different problem, since the tubes are biased to cutoff. When a signal is applied to the grids of this stage, the following happens. During the first half cycle of signal, the grid of tube 1 becomes more and more positive, allowing more and more plate current to flow. As the signal voltage rises still higher, the grid becomes positive with respect to its cathode, and therefore draws current from the electron stream. The grid of tube 2, on the other hand, is driven more and more negative with respect to its cathode. The grid of this tube is already biased so far negative that no plate current can flow, and so this additional amount of bias causes no change in the operation of tube 2. As the polarity of the signal reverses, the roles of the tubes also reverse. It is obvious that in the class-B amplifier, only one tube at a time is operating.

If the tubes are biased to a point even more negative than cutoff, even by a slight amount there will be a point (where the signal is transferred from one tube to the other) where neither tube is handling the signal. This clearly is a form of distortion. Even when the tube is finally conducting, some distortion is present because a rise in grid voltage does not produce a rise (corresponding) in plate current near the region of cutoff.

The point where the signal is transferred from one tube to the other is known as the crossover point, and therefore, the distortion produced at this time is known as crossover distortion.

Nearly all high-fidelity vacuum-tube amplifiers operate at a point somewhere between class A and class B, usually closer to A. This condition is known as class AB_1. Because of this, the topic just discussed would have little more than academic interest to us, were it not for the introduction to the audio field of transistorized power amplifiers, which may contain one or more class B stages. They are used because they are more efficient than class-A circuits, since, when no signal is applied, no appreciable current flows. This greater efficiency leads to cooler operation, which is necessary to prevent excessive heat from damaging transistors.

Crossover distortion is minimized in these circuits by large amounts of feedback which make the base-collector relationship more linear.

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Crossover Networks

Q. I have seen very little published information concerning crossover networks. Would you supply information so that I can tackle construction of them? What is the reason for the use of such networks? Ray E. Roehrick, East Chicago, Indiana.

A. Research has not yet disclosed the perfect speaker. A good speaker should be capable of reproducing the entire audio frequency spectrum flat from 20 to 20,000 cps. Distortion other than frequency discrimination should be under 2 per cent throughout the range. It has been found that if a speaker reproduces the lows well, it cannot vibrate rapidly enough to reproduce the highs with sufficient magnitude. If the size of the speaker is reduced in order to enable it to vibrate at sufficient velocity and amplitude to reproduce the highs frequencies, the unit will lack low-frequency output because the speaker cannot couple to enough air to produce good low-frequency radiation. An 8- or 10-inch model is usually the best compromise.

Most of us dislike compromise and so we have found it advisable to use more than one speaker. Each speaker specializes in reproducing a specific portion of the spectrum. The obvious procedure is to connect all the speakers across the amplifier's output, and each will automatically reproduce its part of the spectrum. Division is automatic, since the speakers cannot reproduce each other's frequencies adequately. This method of connection is a poor one for at least three reasons, however:

(1) It is difficult to get good high-frequency response from even a small speaker unless all moving parts are extremely light. Therefore, many such units (tweeters) are structurally weak. If frequencies below those for which the unit was designed are allowed to enter it, their amplitude would be too great, the elastic limit of the mechanism would be exceeded, and of course, it would be destroyed. Even if the undesired frequencies did not ruin the mechanism, the output at these frequencies would be distorted because of the non-linear mode of vibration which the cone produces when confronted with frequencies which are beyond its capabilities.

(2) The low-frequency radiator (woofer) may have resonances in its upper register which are unpleasant. Therefore, it is best to restrict the range of this speaker so that the speaker will not be excited by energy at its resonant point.

(3) Each speaker placed across the line lowers the impedance presented to the amplifier. When each speaker is assigned a definite portion of the spectrum, this effect is minimized.

It is obvious now that we must divide the spectrum for use with specialized speakers. This division is accomplished through the use of circuits known as crossover networks, or frequency dividing networks. The discussion and circuits following illustrate the manner by which these networks accomplish their purpose. Notice that there is no sharp cutoff or transition at the point where one speaker takes over from the other, High-Q, sharp-cutoff filters would introduce ringing at each transition point, or crossover point as it is mostly commonly termed. 12 db/octave is usually the maximum gradient of attenuation above or below the crossover point, as the case may be. Some engineers advocate a more gradual slope. I personally favor slope between 3 and 6 db/octave. It must be remembered, however, that, with this gradual rolloff frequencies outside the range normally intended will be present in the various speakers comprising the system, and the speakers must be designed to handle more of these undesired frequencies than if they were used with a network having a higher degree of attenuation.

It sometimes happens that the woofer used with a particular speaker system is quite uniform in response up to, perhaps, 5000 cps, after which the highs roll off smoothly. There is, in this case, no need to limit the range the woofer is to handle. Obviously a tweeter must be used to restore the highs. It must not, however, operate to any great extent until 5000 cps is reached, after which point its output rises as that of the woofer falls. *Figure* 1 shows a circuit which could serve this purpose. C_1 is

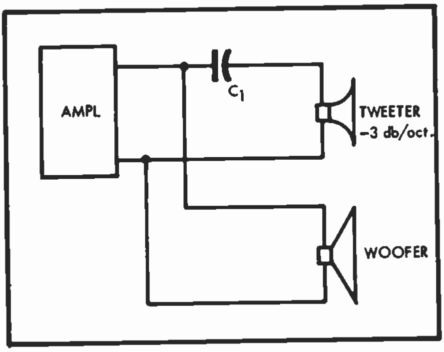

Fig. 1

chosen to have a reactance equal to that of the tweeter at 5000 cps. Below this frequency the capacitor has a reactance which is greater than that of the tweeter so more of the voltage is lost across the capacitor than is developed across the tweeter. As the frequency decreases, less and less power is available to the tweeter, since the reactance of the capacitor becomes larger and larger in comparison with that of the tweeter. As the frequency decreases, the tweeter is gradually isolated from the line. It therefore does not load the circuit when not in use.

Suppose the composition of the tweeter were such that, with the attenuation of-

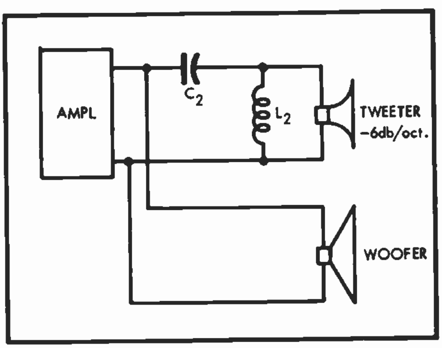

Fig. 2

fered by this simple network, the low frequencies were great enough to cause tweeter damage. The circuit shown in *Fig.* 2 can be used to cause a more rapid attenuation of the lows. This circuit is similar to that of *Fig.* 1, but has a shunted inductance. C_2 functions as before, but its value should be decreased to 0.707 times the value as calculated above.

L_2 is designed to have a reactance at the crossover frequency which is equal to 1.414 times the reactance of the tweeter at the crossover point. As the frequency decreases, the reactance of L_2 increases. This results in the total reactance to the capacitor being less than it would without the inductance across the tweeter, with the result that the voltage division between capacitor and tweeter is greater than the attenuation offered by the circuit of *Fig.* 1.

Suppose now that we have found it necessary to attenuate the high-frequency signals feeding the woofer. We still wish to attenuate the lows fed to the tweeter as before. The circuit of *Fig.* 3 could be used. Notice the similarity between *Figs.* 1 and 3. (Consider the tweeter hookup in *Fig.* 1, and the woofer hookup shown in *Fig.* 3.) In *Fig.* 3, L_1 is in series with the woofer, whereas in *Fig.* 1 C_1 is in series with the tweeter. L_1 is designed to equal the reactance of the woofer at the crossover frequency. As the frequency rises above this point, the reactance rises higher than that of the woofer, and again a voltage divider is formed. Less and less signal is available to the woofer as the frequency increases. As the tweeter takes over, the woofer is gradually removed from the circuit, "unloading" the line except for the tweeter. In all of the preceding circuits, the impedance across the line varies somewhat.

A circuit known as a constant-impedance network is shown in *Fig.* 4. This circuit attenuates the response of the woofer still further. Notice that the circuit for the woofer branch of the network is similar to that of the tweeter branch. Of course, the roles of the capacitors and inductances are reversed, since they behave oppositely with regard to frequency attenuation. This circuit is the standard parallel configuration of a constant-impedance two-way crossover network, and the impedance presented to the amplifier is essentially constant throughout the entire audio range. Values are as follows: $L_2 = 1.414 \times R_0/2\pi f$ and $C_2 = .707/2\pi f\, R_0$ where R_0 is the impedance of the speakers (and the output transformer tap) and f is the crossover frequency. Note that both inductances are of the same value and both capacitors are the same.

In many instances, it is desirable to add a third speaker to the system. The frequency range covered by the woofer is reduced to perhaps 250 cps. The third speaker covers the gap from 250 to 5000 cps, and is therefore known as the midrange speaker. *Figure* 5 shows the complete circuitry for a crossover network which can be used with a three-way speaker system. L_2 and C_2 are connected as in *Fig.* 4, and supply signal to the woofer. Also across the line from the amplifier are C_3 and L_3. Signals for the two remaining speakers are taken across L_3. C_3 is designed to attenuate all signals

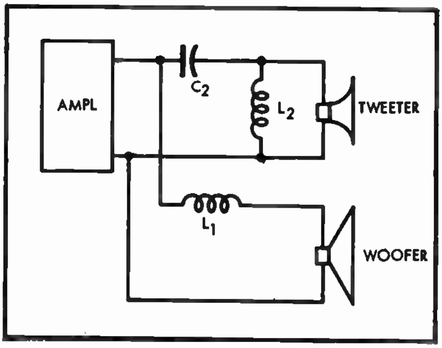

Fig. 3

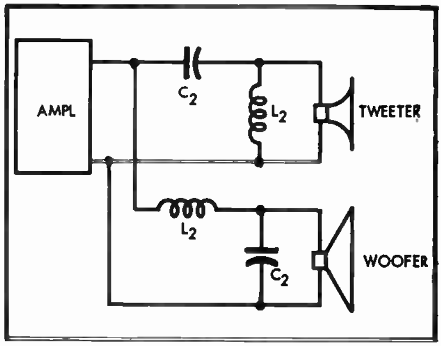

Fig. 4

below 250 cps, as is L_3. From L_3 we use another crossover circuit similar to that of *Fig.* 4. The remainder of *Fig.* 5 is, therefore, a two-way network. The midrange speaker can be considered to be the woofer, and the tweeter to be itself. Since attenuation below 250 cps has already been accomplished by L_3 and C_3, it is necessary only to attenuate frequencies above 5000 cps from appearing in the midrange speaker, and to attenuate those below this figure for the tweeter. Values for C_1, C_4, L_1 and L_4 are calculated as for C_2 and L_2 in *Fig.* 4, using a crossover frequency of 5000 cps, the point where the tweeter takes over. This wiring of the midrange is exactly the same as that of the woofer in *Fig.* 4, except that the values of its associated inductance and capacitor are different from those associated with the woofer. Like the midrange, the tweeter derives its signal across L_3. L_1 shunts the lows around the tweeter, while C_4 provides an easier path for highs than for lows. C_1 and C_4 have the same values, as do L_1 and L_4.

The frequencies chosen for purposes of this discussion are purely arbitrary. Some tweeters are made to take over at 7000 cps, while still others are designed to function at frequencies as low as 1000 cps. There is also considerable latitude with regard to woofers and midrange units.

The circuits presented here are basic, but there are variations. Some networks are designed to attenuate the response at even a more rapid rate than *Fig.* 5. Others are provided with switching provisions, so that a wide variety of crossover points can be obtained. Still others are series circuits; these are more costly to build, but many engineers favor them over the more common parallel constant-impedance networks.

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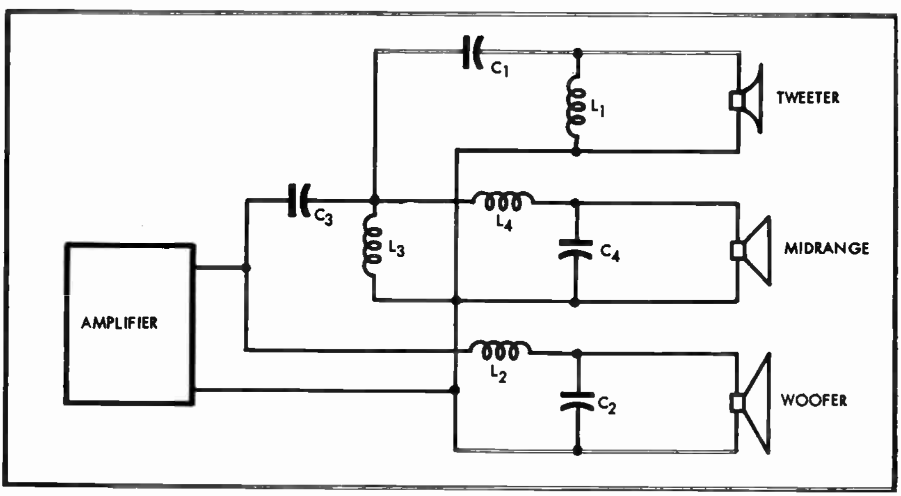

Fig. 5

Speaker Efficiency

Q. Would you please enlighten me as to the relationship between power output of an amplifier and the efficiency of a speaker? For example, I am using a speaker whose efficiency is 10 per cent. How powerful an amplifier do I need in order to drive the speaker sufficiently to obtain 20 watts of musical program level? Can it be that we only obtain two watts output from a 10 per cent efficient speaker when this speaker is driven by a 20-watt amplifier? Fernando Sim, Manila, Philippines.

A. When a speaker has a maximum power handling capacity of 20 watts, it means that 20 watts is the top amount of power which can be fed into the speaker without damaging it or causing serious distortion. If the speaker is 10 per cent efficient, it will when supplied with this maximum of 20 watts, produce 2 watts of acoustical output. Remember that, by definition, efficiency is equal to the power fed into a device, divided by the power output, or yield, of that device. 2 watts output, as in this example, sounds like a small amount of power, but bear in mind that a full symphony orchestra playing at top fortissimo only develops about a watt of acoustical power. Because of this, you will not need a 20-watt program level from your speaker system. (Such a level would most certainly injure your ears, were you to stand near the system when it is giving out with that much sound.) However, assuming that for some reason you do need such high program levels, and further assuming that your speaker system is 10 per cent efficient, you would need an amplifier capable of delivering a power output of 200 watts in order that these conditions be met. Further, you would need a speaker system whose power input capabilities are at last equal to 200 watts, and if this program level is to be maintained over long periods, you would need a system capable of peak power levels of from 250 to 300 watts.

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Speakers and Infinite Baffles

Q. We are considering the installation of an infinite baffle in a wall. Free cone resonance and efficiency are doubtless to be selected with some discretion in this regard. We understand that some of the best speakers for this use are in the relatively inefficient class. How is better bass definition attributed to low efficiency? What bearing has the flux density upon efficiency? What cone resonance will be best for most realistic reproduction in the home? L. B. Osborn, Newburgh, Indiana.

A. Efficiency is governed by the flux density and compliance. The less the compliance and the greater the flux density, the greater the efficiency of the speaker. Because the back wave is lost with infinite baffles, the cone must be free to travel a great distance to make up for this loss. Since low compliance and high flux density both hinder cone travel, speakers having these characteristics will not work well in an infinite baffle. If the cone is too compliant for the amount of rear loading upon it, the speaker may be damaged. For this reason, I like to use a speaker with rather high compliance and high flux density. The flux density limits cone travel after the cone is in high-amplitude motion and it thereby prevents possible damage during power peaks. Poor compliance is a constant restraining force and should be avoided. Unless your baffle is specially designed, I don't recommend a speaker having a very high compliance. You will have lots of intermodulation distortion at best, and you can probably damage the speaker mechanism.

Since an infinite baffle tends to raise the resonant frequency of the speaker cone, it is advisable to use a speaker whose resonance point is as low as possible. As the compliance increases, resonance is automatically lowered. If the infinite baffle is located in a small room, it is unnecessary to have a speaker with a resonant frequency much below 30 cps, since the volume of air contained in the room is not sufficient to reproduce frequencies much below 35 cps. A speaker's 30-cps resonance may be raised easily to 35 cps or higher, depending upon the size of the space behind the cone.

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Cabinet Dimensions

Q. I am thinking of building a large bass reflex cabinet of approximately these dimensions: 2½ ft. × 2½ ft. × 6 ft. I would like to know whether these dimensions are suitable, provided that the port is properly tuned. W. A. Long, Akron, Ohio.

A. In theory, the optimum ratios for the dimensions for a bass-reflex enclosure are 1:2:3. Therefore, assuming the largest dimension of your cabinet to be 6 ft., the others should be 2 ft. and 4 ft., respectively. Because of the large volume of such a cabinet, it will probably be necessary to scale down all dimensions, keeping their ratio intact.

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Infinite Baffles

Q. *Among the objections to the infinite baffle type of enclosure is that the pressure built up inside the cabinet by cone travel interferes with free operation of the speaker, primarily by raising its resonance. How about drilling one or two small holes in the side of the cabinet to relieve this pressure? At the same time, so little of the back wave would escape as to be negligible. Richard J. Galvin, Chicago, Ill.*

A. When a speaker whose suspension is flexible is used in conjunction with an infinite baffle, the speaker is in need of external damping. The pressure built up within the cabinet is a pneumatic spring which performs this damping function. Our own experiments have demonstrated that a speaker having a flexible suspension is the type which works best in such an enclosure, rather than one whose suspension is not very compliant. When the pressure is too low, the speaker will, among other things, tend to bottom more easily, possibly damaging it. Of course, if the pressure is too great the speaker will be over-damped and, therefore, impede the motion of the cone, especially at low frequencies, for it is on low frequencies that the cone travel is greatest. The easiest and most effective way to lower pressure is simply to increase the volume of the cabinet. If the box is large enough, no holes are needed to relieve pressure. If the box is too small, relieving the pressure will probably do little to improve performance, because the cabinet's physical size will not be sufficient to prevent front and rear wave cancellation.

Cable Lengths

Q. What are the maximum lengths of cable which may be attached to a loudspeaker of a given impedance without a loss of more than 0.1 db? J. Kass, Asbury Park, New Jersey

A. There are two variables which must be taken into account in order to answer this question. One is the impedance of the speaker and the other is the resistance of the connecting cable. For a given length of cable, it must be kept in mind that a cable is composed of two separate conductors, each of which contains a specific resistance per unit length. The resistance is inversely proportional to wire diameter. The following table shows the length of a cable used for connecting speakers to amplifiers with impedances ranging from 4 ohms to 500 ohms. The table represents the actual physical length of line, rather than the lengths of the two conductors laid end to end.

TABLE I

Wire Guage	Line Impedance in Ohms				
	4	8	16	150	600
22	20′	40′	80′	800′	3200′
20	30′	60′	120′	1000′	4000′
18	50′	100′	200′	1600′	—
16	80′	160′	320′	2400′	—

Distance from Crossover Networks

Q. *My question is: How far can speakers be operated from their crossover networks, assuming that fairly large wire is used? If the distance over which this operation is permitted is considerable, say 40 feet or more, then why should one not use the same crossover network for a number of speaker combinations? W. E. Dancey, Houston, Texas.*

A. In this column, Jan., 1958, there is a table showing lengths of line *vs.* wire diameter, *vs.* impedance for 0.1-db loss. By using this table as a guide, negligible distortion of the characteristics of the crossover network will result. This is because the resistance of the line is negligible as compared to the nominal impedance of the circuits involved.

One word of caution: If the crossover network is of the non-adjustable type, be sure that all the speaker systems to be used require the same crossover point. If the speakers are to be switched (and this is certainly the best way of handling this project), be sure that the switch contains sufficient positions, one for each speaker system, and enough poles, one for each element of a speaker system.

Balance

Q. I have a fine amplifier in most respects, but I do notice that, as the gain is reduced, the balance between highs and lows is not of the same proportion as when the gain is turned up. There seems to be a very definite point at which the frequency becomes good. What is this condition and how can it be corrected? Name Witheld, Lake Worth, Fla.

A. This sounds as though it might be an advanced case of Fletcher-Munson trouble, and the effect is due to the lowered sensitivity of the human ear to low frequencies as level is reduced. On a perfectly flat reproducing system, music will appear to have the proper balance only when the reproduction level is identical with that of performance. If this is the problem in your case, you might find a solution in installing a compensated type of volume control in place of the "flat" or uncompensated control now in use. There are other causes which could result in a loss of highs as the volume is lowered, the most common of which is due to high stray capacitance in a grid circuit. As capacitive reactance at the higher audio frequencies approaches the input impedance of the amplifier, the amplitude of the high frequencies is diminished with respect to middle and lower frequencies. One form of such capacitance is that between the resistance element of a volume control and its case. This varies with the setting of the control, so that when the control is in the most advanced position, the effects of the shunt capacitance are at a minimum. The larger the resistance of the control, the smaller will be the shunt capacitance needed to cause this high-frequency loss. To minimize this effect, replace the present volume control with one whose resistance is no more than half that of your present control. By whatever amount this resistance has been decreased, the value of the coupling capacitor feeding this control should be increased, so as not to impair the low-frequency performance of the amplifier. If the value of the coupling capacitor is not increased to meet the decreased control resistance, losses will occur because of the voltage divider formed by the reactance of the capacitor and the resistance of the control. As frequency decreases, the reactance of the capacitor increases until finally there comes a point where that of the capacitor equals that of the volume control. As frequency proceeds below this point, it is obvious that more voltage will be lost across the capacitor than across the control. From this point on, low frequencies will be attenuated at the rate of 6 db per octave. Æ

Power Output

Q. I have been checking the power output of my Williamson-type amplifier and have had some disappointing results. I am using 6L6-GB tubes as power output stages, driving a UTC LS63 output transformer, right into an 8-ohm speaker. The tube manual states that these tubes, operated with cathode bias with 360 volts on their plates and 270 volts on their screens, would deliver 24.5 watts in class AB. I'm operating the tubes at 340 volts on the plates and 250 volts on the screens, and with a 250-ohm cathode resistor, bypassed with 250 μf. The grids are wired with 470,000-ohm resistors. If I disconnect the driver stage from the 6L6 grids and feed it into separate 470,000-ohm resistors, I can obtain 15–16 volts RMS to drive the grids of the 6L6's. This waveform is clean as seen on a 'scope. However, as soon as I connect the driver to the 6L6 grids can only drive them to 10 volts RMS. Beyond this point, the waveform flattens off. At this point I can get only about 10 watts out. Can you explain why I cannot approach at least the 20-watt level with the above parameters? W. E. Henry, Albuquerque, New Mexico.

A. It is logical to expect that, when the output of your driver stage is connected to a dummy load, more output would be obtained than when the same driver was connected to its proper grid load, with the remainder of the circuit made operative. This is because the output stage is designed to feed back a certain amount of its energy in such a direction as to cancel a portion of the driver's output voltage. A driver stage of higher capacity would have to be used to overcome this difficulty. Such a stage might take the form of a push-pull parallel configuration.

Think back to the original Williamson circuit, in which 807's were used. In some circuit arrangements as these tubes are easily capable of delivering 40–50 watts output, and hams use them as Class B linear amplifiers and easily obtain 100 watts from them. Williamson got 8 watts out, using them as triodes, but with very low distortion. Æ