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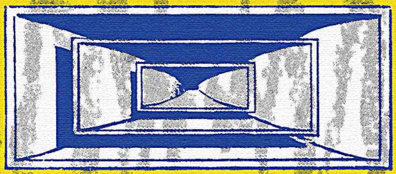
AUDIO SYSTEMS HANDBOOK



BY NORMAN H. CROWHURST

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By Norman H. Crowhurst

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Preface

Audio is a many-faceted subject, although in terms of employment it's one of the smaller vocations. This means the need for literature is seldom adequately supplied, because publishers have to live, too. They are reluctant to publish a book for a mere handful of people, because its difficult to make money—or even break even—that way. So a book like this one must try to condense the most used information into its most assimilable form, so it will be of the greatest possible use to the biggest possible proportion of the still rather small number of people in this interesting business.

And an interesting business it is. Having grown up with it, the author has seen some crazy things happen—the kind that makes one comment that truth is stranger than fiction. And there's a zeal about audio people that one doesn't find anywhere else. But still, not enough is written. This book fills what the author hopes will be a substantial portion of that void. It bridges the gap between the non-technical treatment, where an audio system is accepted as an entity designed by experts in their field, and the engineering treatise that may never be published, but which these experts carry around in their heads!

In other words, it's for those people whose job is in audio systems, or who may want to "dabble" in them, and who haven't found some of the answers they need. May it help them to contribute usefully to that vital human necessity—audio communication between the genus homo sapiens!

Norman H. Crowhurst,
Gold Beach, Oregon

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Chapter 1

Amplifiers

The central feature of any audio control system is at least one amplifier. The type of amplifier chosen, or where more than one is used, their number and type, depends on the application to which each is applied. In simpler systems, such as a public address installation for a community hall or meeting place, a single amplifier is enough. While it should provide for all likely contingencies, it also should be simple to operate, and flexible enough to meet those needs.

Most such systems need a microphone. But to the uninitiated, a mike is just a mike. If you ask "what kind?" they may tell you something irrelevant to the intent of your question, such as how much it cost or where they bought it... an answer that doesn't help you to determine whether or not it can be used with the amplifier they have.

The uninitiated user doesn't understand impedance. His experience may tell him that some microphones are more sensitive than others, but he cannot understand why one microphone is more sensitive on one amplifier and practically useless on another, while a different microphone behaves exactly vice versa. To a person who doesn't understand what impedance means, that experience just doesn't make sense. He probably begins to realize that you must use a mike whose impedance is the same as the amplifier input, or it doesn't work as it should, which is a step in the right direction. But if he's going to learn everything the hard way, he has a tremendous lot to learn!

He's going to run into hum pickup problems, sooner or later. Different impedances require different measures for effective shielding against hum. Our uninitiated friend may try what worked very well once before, without realizing he was then dealing with a different impedance, and find that now the same "trick" is useless.

At the output end of the amplifier, the uninitiated user accepts a speaker as a speaker. It may be big or small, indoor or outdoor. After acquiring a little more sophistication, he may know some are more directional than others, but impedance...What's that? In connecting speakers to a system, a precisely correct impedance can be more important than it is at the front end, with microphones and other program sources.

He encounters all that in a simple system with only one amplifier. So far, he needs to know only that an amplifier provides that rather vague quantity—"amplification." But when he uses more amplifiers in a more complicated system, he'll have to learn that he needs the right "kind" of amplification to suit each place.

AMPLIFICATION

Amplification is relative, and db level is relative—facts that confuse the beginner. Saying an amplifier has so many db gain means its output will be that many db higher in level than its input. But because level also possesses an absolute scale, any particular amplifier is suited for raising the level only at an appropriate part of that scale.

Clouding the situation even more is the fact that level is never constant. "Program" may have a peak level, which the system should handle without distortion. The average program level will invariably be less than this peak level. And some of the nuances of the program also will be considerably lower than the average level. These, too, should be audible.

Were our uninitiated friend to use as a microphone preamplifier a power amplifier intended to accept "line level" input and deliver power output to drive speakers, he'd probably end up with so much hum and noise that no program would be audible against it. Yet this same amplifier operated under circumstances for which it was designed might be completely silent, no hum, no noise, just a faithful reproduction of the program it handles. Conversely, if he tried the opposite—to use a microphone preamplifier as a booster to feed some additional loudspeakers—(after all, it does amplify)—he'd get nothing, because it's designed output doesn't even approach 1 watt, and the speakers may need several watts to drive them.

ABOUT db AND IMPEDANCE

Amplification is almost universally quoted in db (decibels). But this gain figure is not of much use without specifying both input and output impedance. Further, with all these items of information, input and output work effectively only over a certain range of levels.

Gain given in decibels is purely relative—output is so many db higher than the input. If one millivolt input produces two volts output, then two millivolts input produces four volts output, and so on. However, this is true, both up and down, only to certain limits—that's why levels are so important.

Levels are specified in db in relation to some reference level. The commonly accepted reference level is 1 milliwatt, and levels stated with this as reference are called "dbm." Thus -30 dbm means a level of -30 db referred to 1 mw, which happens to be 1 microwatt. Or + 40 dbm means a level of +40 db referred to 1 mw, which is 10 watts. The reference for all db figuring, whether absolute or relative, is basically in power units. Thus, 10 db represents a power ratio of 10:1, 20 db is 100:1; 30 db is 1,000:1; and so on. Lesser level ratios use 10 times the logarithm to base 10 of the ratio. Thus, between 1 and 10 the power ratios and equivalent db figures are related in the manner tabulated in the chart.

All this is quite neat, but for the lower levels of amplification another reference is often used, which seems more meaningful there. What matters from the operational viewpoint is not so much the energy content as the voltage (or current) level. Power drives loudspeakers, but microphones, pickups, tape heads, and so forth are usually regarded as voltage "generators" rather than power "gen-

Power ratio	db	db	Power ratio
1.5	1.75	1	1.26
2	3.01	2	1.585
2.5	4.0	3	2.00
3	4.77	4	2.51
3.5	5.44	5	3.16
4	6.02	6	4.00
5	7.0	7	5.00
6	7.78	8	6.31
7	8.45	9	7.95
8	9.03		
9	9.54		

erators." So at lower levels we are more concerned with amplifying voltages than with the associated power levels.

Even so, the real reference, even at these levels, is power, because we must take impedance into account. First, notice what this means at a relatively high level, where power has a meaning we can comprehend better. Suppose we use that level of +40 dbm, which is 10 watts. If we have 10 watts of steady signal in a 16-ohm loudspeaker, this power consists of about 12.65 volts RMS at 0.79 amps. These values satisfy both measures: for impedance, 12.65/0.79 = 16 ohms; for power 12.65 x 0.79 = 10 watts.

Starting from power and impedance, voltage and current are calculated by:

$$V = \sqrt{PR} \qquad (1)$$

and:

$$I = \sqrt{P/R} \qquad (2)$$

In this example:

$$V = \sqrt{10 \times 16} = \sqrt{160} = 12.65 \text{ volts}$$

and:

$$I = \sqrt{10/16} = \sqrt{0.625} = 0.79 \text{ amps}$$

The same power in a 40-ohm loudspeaker consists of 20 volts at 0.5 amp, which again satisfies both measures; for impedance, 20/0.5 = 40 ohms; for power 20 x 0.5 = 10 watts. Or, if we take the same power at the plate circuit of an old-fashioned tube-type amplifier, which might have a load impedance of 10,000 ohms, the same level calculates to 316.2

volts at 31.62 milliamps. So the same power level, 10 watts in this case, can represent a variety of voltage and current combinations, according to impedance.

At low levels the commonest impedance is a line impedance, which may be 125, 250, or 500 ohms. At zero dbm the corresponding voltage levels are 0.3535, 0.5, and 0.707, respectively. Suppose a level is specified as -55 dbm, actually 3.16 millimicrowatts. So -55 dbm is 0.63 mv at 125 ohms, 0.89 mv at 250 ohms, or 1.26 mv at 500 ohms.

In amplifying at these levels it is easier to "think" in terms of voltage. So we may use voltage ratios as a basis for db gain instead of power ratios, provided the impedance is the same at both input and output. Actually, this is one of the reasons for thinking in voltage—the impedances often aren't strictly the same: impedance is nominal in these circuits, so the nominal power is virtually meaningless.

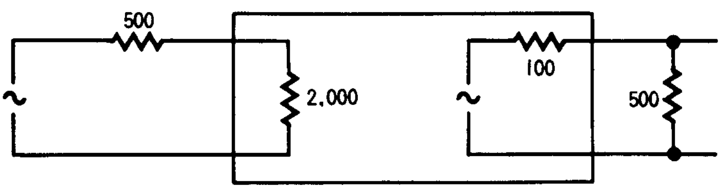


Fig. 1-1. Input and output impedance each have a dual significance—one internal and the other external.

A line amplifier with a designed 500-ohm input and 500-ohm output may have an input impedance of, say, 2,000 ohms and an output impedance of 100 ohms (Fig. 1-1). The purpose of these apparent inconsistencies is to make the voltage gain achieved less dependent on the exact external impedance value connected and also, possibly, so frequency response and other performance details do not depend critically on the impedance values of external connections. But these safeguards, in turn, get us into some inconsistencies that can be confusing.

INSERTION GAIN

First, we must explain the meaning of "insertion gain," as applied to line amplifiers (an amplifier with the same nominal

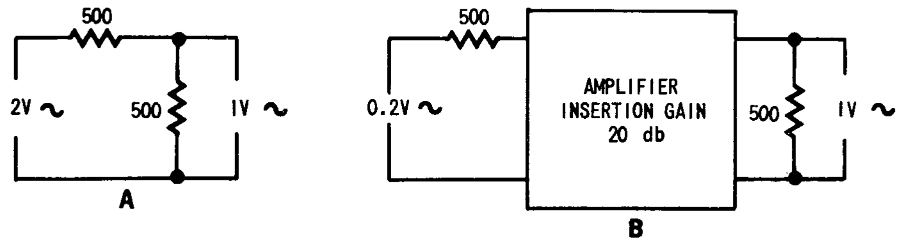


Fig. 1-2. Sketch illustrating the effect of "insertion gain" (see text).

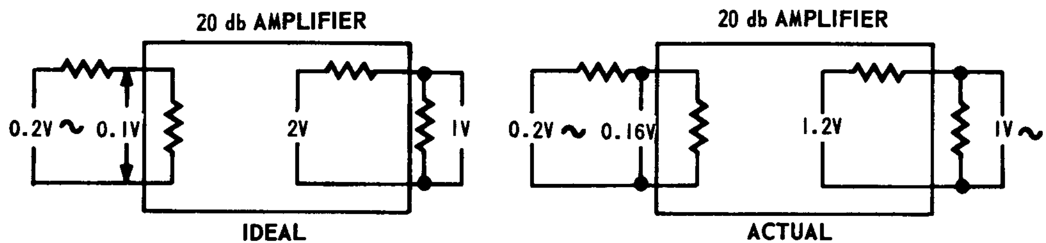


Fig. 1-3. Comparison between the ideal line amplifier and a typical practical form.

input and output impedance). The definition assumes a reference condition in which a voltage source feeds through its own internal impedance (the rated value) into the same load impedance (Fig. 1-2A). Here the voltage transferred to the second (or load impedance) is exactly half the open-circuit voltage from the input. Now, we separate the two resistors and put the amplifier between (Fig. 1-2B). How much higher is the level at the output load than in the first connection? Conversely, how much lower can the input voltage be, so that the output voltage remains the same? Either of these ratios, converted to db, is called the insertion gain of the amplifier. Insertion gain thus indicates how much higher the level becomes as a result of inserting the amplifier.

In the days before certain standards organizations insisted on this definition, the "gain" of an amplifier could mean different things. Often it was the simple ratio of input voltage to output voltage, converted to db. At first sight, you'd probably think this was the same thing, but it isn't. Let's assume the actual input impedance is 2,000 ohms and the actual output source impedance is 100 ohms. In this case, the first figure is what makes the immediate difference. Suppose the insertion gain is quoted as 20 db. This means the voltage change, as a result of inserting the amplifier, is 10 times.

In Fig. 1-2A, two volts input produces one volt on the output resistor. Now insert the amplifier, and the 0.2-volt input will produce one volt on the output resistor in Fig. 1-2B. We know this by the definition of insertion gain. Without better knowledge, we'd imagine the input voltage at the amplifier terminals is 0.1 volt. But 0.2 volt is feeding through a 500-ohm resistor into a 2,000-ohm input, so the input actually receives 4/5 of the 0.2 volt, or 0.16 volt (Fig. 1-3). So, measured the old way, 0.16v input produces one volt output, which would be a gain of only a little over six times, or about 15.6 db. If we took an amplifier with 20 db gain by the old measure, the insertion gain would be more than 20 db.

When the term "insertion gain" first came into vogue, some of the old-timers called this "cheating," but really it isn't! They didn't recognize the true significance because they were thinking only in "voltage" while, actually, the amplifier is

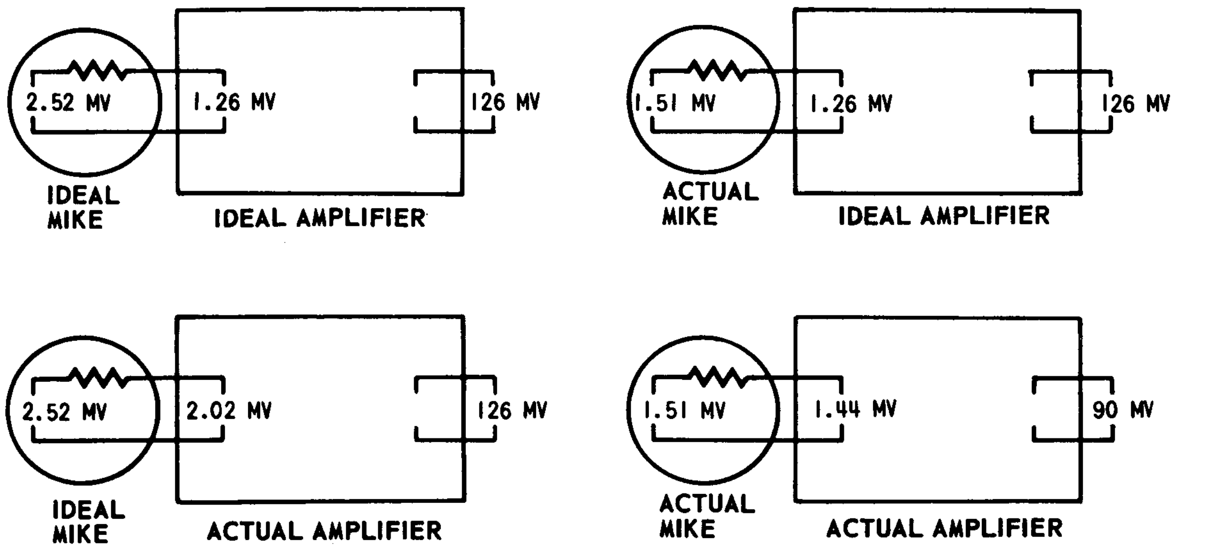


Fig. 1-4. Step-by-step comparison between an ideal transducer (microphone) feeding an ideal amplifier, and a practical combination, developing the difference from ideal insertion gain that occurs in practice.

making an impedance change, as well as raising the voltage. Suppose an "amplifier" didn't increase voltage at all, but instead produced the same voltage across a 100-ohm output as was presented at the input across 2,000 ohms. This would be a gain of 13 db. Although voltage isn't increased, current is, and that is an increase in power and thus gain.

At one time, a stage called a cathode follower (in transistors it's an emitter follower) was called a "gainless" stage. Rather than providing gain, thought of as an increase in signal voltage, it served as an "impedance changer." But providing the same signal voltage across a lower impedance is also a form of gain. A true impedance change—without gain—affects both voltage <u>and</u> current, maintaining the same power level. This is provided by a transformer.

While the use of insertion gain as a specification comes much closer to predicting the correct performance, it sometimes errs a little, particularly where neither of two impedances connected together possesses the nominal value.

<u>Example</u>: The output level rating of a microphone or pickup, designed for connection to a 500-ohm input, is based on its being terminated by a 500-ohm load. Connecting it to a nominal 500-ohm input that presents an actual load of 2,000 ohms will not result in as much gain as the combined specifications predict.

Suppose, based on an assumed 500-ohm load, the microphone is rated at -55 dbm output for an acoustic input of 10 millibars. As calculated earlier, this would be 1.26 mv across 500 ohms. But suppose the actual impedance of the microphone or pickup is only 100 ohms; it is rated to work into a 500-ohm load for purposes of achieving optimum response rather than maximum signal transfer. So the open-circuit signal with a 10-millibar acoustic input would be 1.51 mv not the 2.52 mv that academic theory would suggest (Fig. 1-4). Now suppose this is connected to the input of an amplifier with an insertion gain of 40 db. According to the combined ratings, if we had no knowledge of actual impedances, we would expect the amplifier to deliver 126 mv into a 500-ohm load.

But to achieve this 126 mv at the output, the input should be 2.52 mv (open circuit) with an actual 500-ohm source. Then

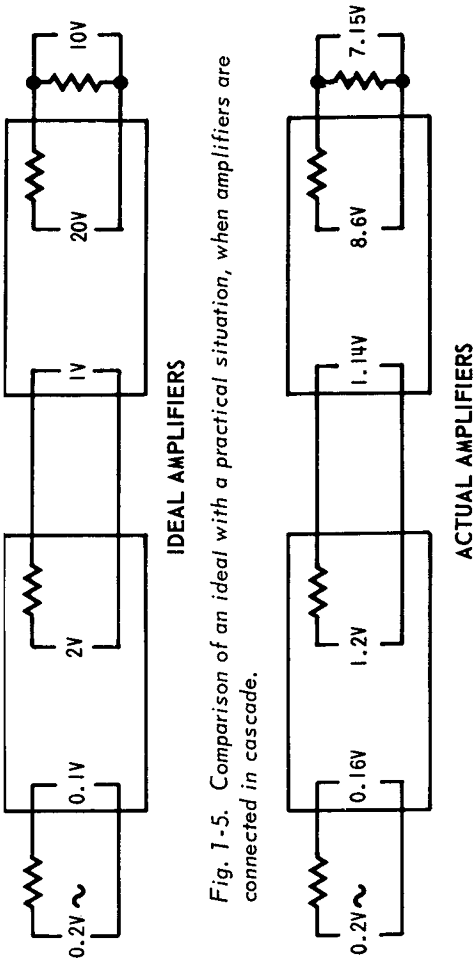


Fig. 1-5. Comparison of an ideal with a practical situation, when amplifiers are connected in cascade.

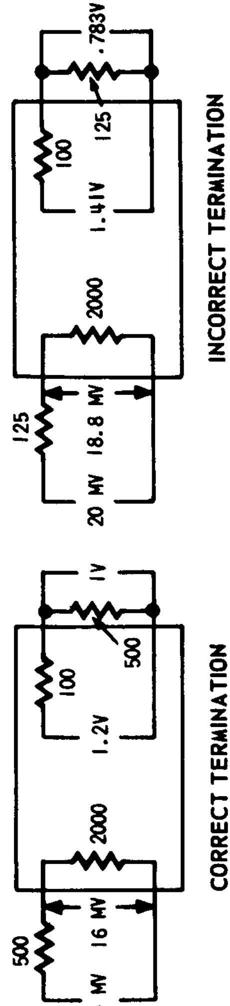


Fig. 1-6. The effect of loading a practical amplifier with external impedances other than that for which it was designed is illustrated here.

the actual input voltage at the amplifier terminals would be 4/5 of 2.52 mv, or 2.02 mv. As it is, the actual input voltage from the microphone will be 20/21 x 1.51 mv, or 1.44 mv.

So the actual output will be 1.44/2.02 times the anticipated 126 mv, or about 90 mv. Thus the overall gain fails to come up to expectations by about 3 db. Precisely the same thing would happen, if the source is a preamplifier output rated as 500 ohms, which really has a 100-ohm source resistance. For example, if two such "voltage amplifiers," each rated at 20 db gain, were connected together, the total gain would be about 37 db (Fig. 1-5).

Now assume that an amplifier is used for a different impedance than the one for which it is rated, but that the external impedances are correct. Suppose it is rated to give 40 db gain with impedances of 500 ohms, and has the same internal impedances just mentioned, but is actually used with external impedances of 125 ohms. This means that, with 500-ohm input and output, the input voltage would be 20 mv for an output of one volt. The input at the amplifier terminals would be 16 mv (not the 10 mv imagined) and the open circuit output voltage would be 1.2 volts (not the two volts imagined). So the voltage gain, from input terminals with the output open-circuit, is 75, or 37.5 db.

Now insert the same 20 mv through a 125-ohm input resistor, and with a 125-ohm output load (Fig. 1-6) the input voltage will be 2,000/2,125 x 20 mv = 18.8 mv. The open-circuit output voltage would be 75 x 18.8 mv = 1.41v. Loading this 100-ohm source with a 125-ohm load will yield an output voltage of 125/225 x 1.41 = 0.783v, instead of the one volt with the correct termination. So changing the terminations has reduced gain by a factor of 0.783, or about 2.1 db. As a matter of interest, let's compare that with the theoretical model on which calculations are based, where the amplifier has actual internal impedances to match the external ones. In this case, the input of 20 mv would be attenuated to 10 mv at the amplifier terminals, to be amplified to an open-circuit output of two volts, loaded down to one volt by connecting the matching load.

Now change impedance at both ends to 125 ohms. The input changes from 10 mv to 500/625 x 20 mv = 16 mv, so the

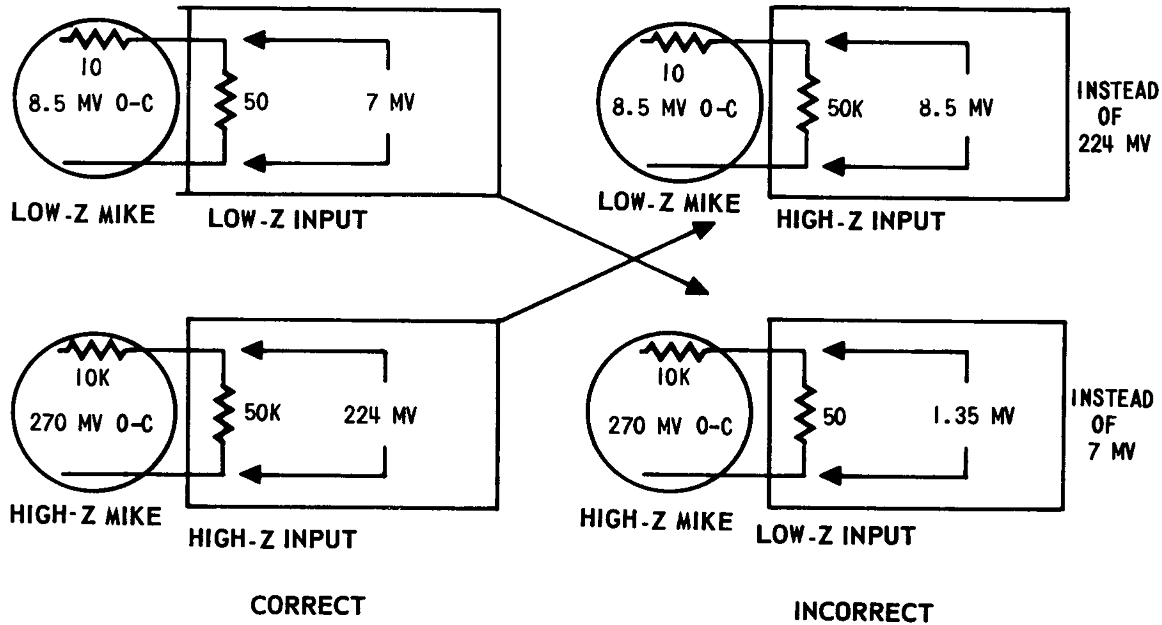


Fig. 1-7. Emphasized here is the effect of connecting microphones of incorrect impedance to opposite kinds of amplifier input impedance.

open-circuit output changes from two volts to 3.2 volts. But this is loaded down to 125/625 x 3.2v = 0.64v. This is a loss of gain, due to mismatching at both ends, of almost 4 db, very nearly twice as much as in the actual amplifier just discussed. So an advantage of this design is reduced susceptibility to precise matching.

SPEAKER vs AMPLIFIER OUTPUT IMPEDANCE

We mentioned high- and low-impedance microphones earlier. Some microphones come with low impedance, some with "line" impedance and some with high impedance. Some are switchable between two or maybe three different impedances, so they can always be adapted to suit the amplifier input. Some are designed to operate at only one impedance. If you connected a low-impedance microphone to a high-impedance input, the signal voltage it provides will be almost 30 db lower than it should be for that type of input. On the other hand, if you connect a high-impedance microphone to a low-impedance input, you will virtually short-circuit the microphone output, and it will provide about 14 db less signal current than the input needs (Fig. 1-7). More importantly, in this case, the relative short-circuit loading will spoil quality.

In power amplifiers intended to deliver power to speaker systems, the situation is a little different, but in this, too, amplifier design attempts to provide more latitude than theory would indicate. In this instance, using a lower output source impedance serves to provide better damping for loudspeakers and more constant output voltage as the number of speakers connected may be changed.

Suppose an amplifier is rated to deliver 100 watts to a speaker load of 16-ohms impedance. By equation (1) this calculates to an RMS voltage of 40v. What does this 40-volt output mean?

First, a nominal load of 16 ohms, made up of speakers so that the load calculates to this value (which is dealt with in detail in Chapter 3), will have similar actual impedance characteristics to those of a single speaker, typical of those connected. The manufacturer rates his speakers, in impedance and power, according to nominal values—say 40 ohms, 10 watts, for example. This does not mean an actual electrical power input of 10 watts at every audio frequency it handles, but what would be 10 watts if the same voltage were applied to an actual 40-ohm resistor. Thus, to get the rated 10 watts, 20 volts must be applied to its terminals. At some frequencies the impedance may be higher than 40 ohms and reactive, so the actual electrical power it receives is far less than 10 watts. But its performance at a nominal 10 watts is based on applying 20 volts to its terminals at all frequencies.

To ensure that the amplifier works consistently with such variations in impedance, its internal impedance is invariably made much lower than its rated value, which names the load that should be connected. Then the internal design is such that maximum output is really a voltage, rather than a power. Above that certain voltage output, distortion begins to increase rapidly. And the voltage at which distortion begins to show does not depend too much on the load impedance into which output power is fed, provided this is higher than the nominal value. So a 40-volt maximum output into an actual resistance load of 16 ohms ensures that the amplifier will provide the power to correspond with what the speaker manufacturers rate as 100 watts, for driving a load of suitably matched loudspeakers.

If the impedance match is wrong, some output power will be sacrificed. If the nominal impedance of the speakers connected is higher than the nominal load for the amplifier, the amplifier still delivers the same undistorted output <u>voltage</u>, so connecting a speaker load making up a nominal value of 32 ohms to a 100-watt amplifier's 16-ohm output terminals will reduce the nominal available output to about 50 watts, maybe a little more.

Connecting an impedance of too low a value is less predictable. At some frequencies speaker impedance actually equals the rated value. At such frequencies, the amplifier will be drastically overloaded. Distortion may become serious at 1/4 the rated power. But at other frequencies, where the speaker impedance is twice its nominal value, the same amplifier will deliver twice the actual power it would be expected to provide for the same speakers at those frequencies.

The preceding cases exemplify the whole story. Obviously, with the latter mismatch the speaker output will sound much louder, but more distorted, at least on certain frequencies. The former mismatch will cut power about in half, which is only 3 db, and not very noticeable, unless there happens to be a background noise problem.

LEVEL LIMITATIONS

As we have already implied, upper-level limitations are usually determined as a maximum voltage. This is true both at line amplifier levels and with power amplifier outputs. Low-level limitations are determined by signal-to-noise ratio. Some latitude can be allowed, according to the margin by which the minimum signal level should be above noise.

For communications purposes, where receiving an intelligible message is the primary requirement, the margin need be only relatively small. In fact, the compression of signal level can assist. This may also help in large systems where achieving adequate power for high signal levels presents a problem when combined with weaker signal components to override background noise, whether acoustic or electrical. But for quality reproduction, where dynamic range is desirable to the impression of fidelity, a wider margin over background noise is necessary.

One more point must be observed relative to level ratings. Suppose an amplifier designed for 16 ohms has to feed only a single 16-ohm speaker, and the amplifier's rating is 100 watts. The speaker may be rated at only 10 watts, but the impedance is correct, so one may be tempted to make a direct connection and keep the amplifier "turned down." This is dangerous. The amplifier is always capable of delivering that 100 watts, which would probably destroy the 10-watt speaker.

Although no foreseen circumstance could drive the amplifier to its maximum output, it is still dangerous. Some fault may develop in the attenuator that prevents the amplifier from developing full output, or loss of a ground could cause 100 watts of hum to be fed into the speaker. But even if such a fault does not develop, such an operation has a continuous disadvantage. The background noise level of the amplifier always refers to its maximum output. In an amplifier rated at 100 watts output, the background level is inherently 10 db higher than in one rated at 10 watts output.

When this is distributed among, say, ten 10-watt speakers, the background level is no higher than from a 10-watt amplifier feeding one speaker. So, even if the 10-watt speaker is operated safely from the 100-watt amplifier, background noise would be unnecessarily high—by 10 db—because the one speaker would be "getting it all." Under such circumstances, the best plan is to attenuate the power. Provide resistance loads that will dissipate the 100 watts of maximum power, and at the same time provide the speaker with its rated 10 watts.

To calculate this, 10 watts at 16 ohms is 12.6 volts 0.79 amp. 100 watts at 16 ohms is 40 volts, 2.5 amps. So a series resistance to take 40 - 12.6 = 27.4 volts at 2.5 amps, with a shunt resistance to take 12.6 volts at (2.5 - 0.79) = 1.71 amps will serve (Fig. 1-8). The values required are 11 ohms, 70 watts (round figures) and 7.5 ohms 25 watts.

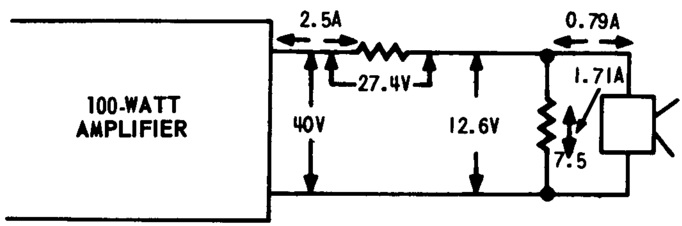


Fig. 1-8. Drawing showing how to absorb excess power when a high-power amplifier feeds only one speaker.

GROUNDING AND SHIELDING

These aspects can become problems particularly in larger systems using several amplifiers. Grounding and shielding are important for two purposes: holding down the pickup of hum fields from the AC supply, which permeate all buildings wired for power; and avoiding instability. If your supply happens to be DC this doesn't obviate the hum problem. It may even make it worse. Supplies of DC usually come either from rotary converters that have brushes, and thus produce a ripple of much higher frequency than that of an alternating supply, or from a rectifier. If the rectifier is single-phase, the frequencies it contains are ripple frequencies starting at twice the unrectified supply frequency. If it is multiphase, a much higher frequency ripple may be far more audible.

In any event, with a DC supply you need an inverter to supply your system with AC power, which may add to your grounding problems. Not that you'll have trouble getting "enough" grounds. The problem is more likely to be "too many!" The important thing in all shielding and grounding connections is to keep all possible high-level ground currents out of low-level ground connections. All amplifiers have high-level signal currents associated with the output, which flow from the supply. Signal voltages are decoupled, but signal currents remain, and these can cause trouble in residual common impedances, resistance or inductance of ground connections, etc.

So the internal connections of an amplifier (which readers of this book should not have to bother with, but it's well to know, in case) have ground returns grouped in such a way that signal-return currents from higher level stages never pass through the ground points of lower level stages (Fig. 1-9). This means that the chassis ground is connected to the circuit ground at the return point of the lowest level stage in the amplifier—the input. This procedure avoids both instability and the pickup of unwanted hum due to incorrect grounding. In any system, the ground returns should likewise be connected as a system, running from input to output in the same way as the signal. If all the chassis bolt to a common rack frame, such a ground will not be necessary. It is sufficient that each internal circuit is grounded to chassis only at its input.

The final source of unwanted pickup, particularly of hum, is in the connecting leads that run at low level, such as microphone cables. Here the variety of hum pickup varies according to the impedance used for the connection, which depends on the impedance of the device connected. Microphones and other source devices have basic impedances inherent in the unit. Crystals or ceramics are high impedance, but of a different kind than other high-impedance devices. Magnetic type devices use a coil which may be wound to any impedance, from low to high. Dynamic, including moving-coil and ribbon mikes, are inherently low impedance and must use a

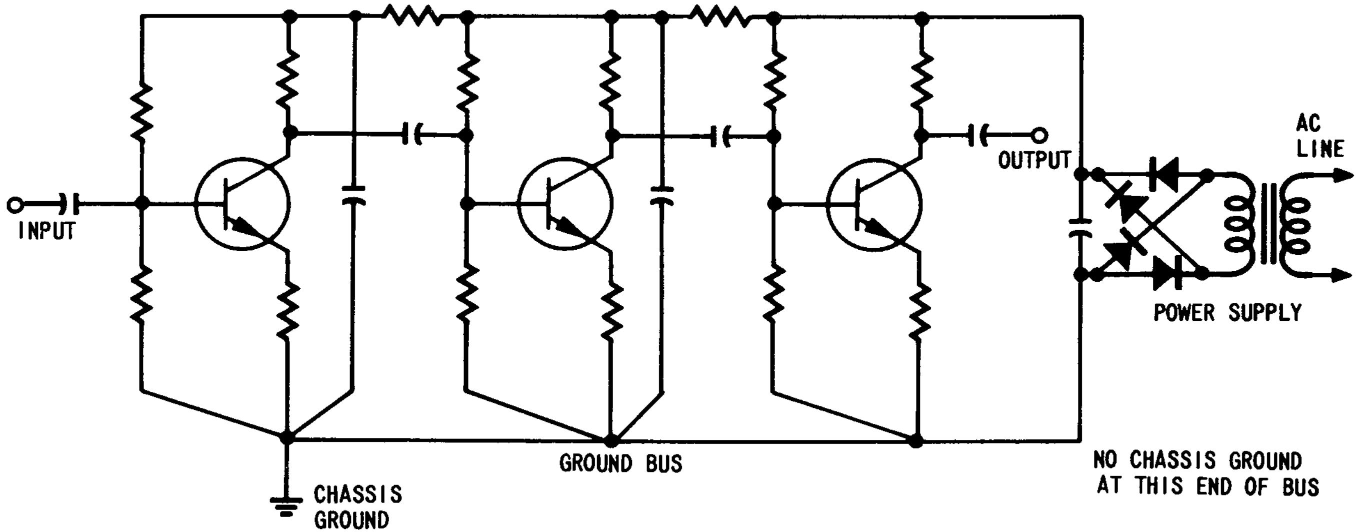


Fig. 1-9. Grounding arrangement in a typical voltage amplifier, correctly designed to avoid ground problems.

transformer to achieve line or high impedance. All devices except ceramic may use transformers internally to change impedance to suit a particular application. But the ceramics are a class on their own.

In low-impedance input circuits, the main form of pickup is inductive—current induced by magnetic fields. As an electric shield does not stop a magnetic field, it is ineffective against this form of pickup. So a single-conductor cable with a concentric sheath is not effective at low impedance, because magnetic fields will still induce currents in the inner conductor that will appear added to the wanted input signal as hum.

To prevent magnetic induction in low-impedance circuits, both connections ("live" and "return") should be twisted together so any induction is neutralized. This is very important. The use of twin shielded cable is of little help, unless the twin is twisted inside the shielding (as it invariably is, these days, because straight twin shield has little use). If you find a piece of twin shielded that isn't twisted inside, it can cause hum pickup when used on a low-impedance connection. Unshielded twisted flex will pick up less hum on such circuit, a fact that can come as a surprise the first time you discover it!

For high-impedance circuits, electric (formerly called "electro-static") pickup is the main cause of hum. You can tell the difference by the sound or frequency content. Inductive pickup usually emphasizes the lower frequencies, as a boomy sound. Electric pickup emphasizes the higher components, radiated by devices that "strike," such as fluorescent lamps, rectifiers, etc. You would describe it as a "ticky" hum.

This is safeguarded by a properly grounded electric shield. The important thing here is to provide a path to ground for any induced hum voltages that do not flow through any signal circuit. If you use a two-core shielded cable, use one internal conductor for "live," one for ground at the end remote from the amplifier input, and one for the shield. Ground the shield only at the amplifier end (Fig. 1-10). If you use single-core cable, you have no problem, but make sure the ground carries through.

For line impedance both forms of pickup are about equally likely to cause trouble, but much less than each one with low-or high-impedance connections respectively. So the connecting cables need to provide protection against inductive and capacitive pickup. They should be twin-twisted shielded. Best results are obtained by using balanced circuits, so that both pairs of the twin are equally "live" but of opposite polarity. Then the ground connection is carried through the shielding (Fig. 1-11).

This form of connection can operate at a very much lower level without picking up hum than either low- or high-impedance connections. It is also much more costly, when you take into account the added cost at the ends, for balanced transformers or other input circuits, and for balanced

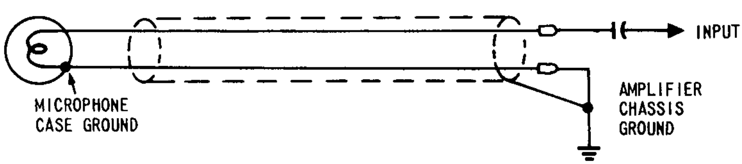


Fig. 1-10. Grounding arrangement with a high-impedance input, using twin-shielded cable.

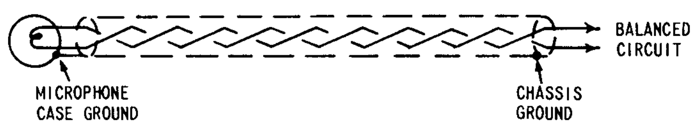


Fig. 1-11. Arrangement for using twin-shielded cable with line-impedance connections.

mixers, as compared with the unbalanced, as well as the higher cost of connecting cable. But where you are working with circuits that connect to long telephone lines, the use of balanced connections is vital. The level on the telephone line is usually kept high enough to avoid spurious pickup, in conjunction with the design of the line to neutralize what pickup does occur. But at the termination, where mixing occurs, precautions are necessary.

The choice of impedance is usually linked with the length of connection likely to be used. Ordinary high impedance (from high-impedance magnetic devices, or any of the ones that obtain high impedance with an internal transformer) can feed only a relatively short distance, using high-quality, low-capacitance single-core shielded cable. About 20 to 30 feet should be considered a limit. Beyond that, serious high-frequency losses begin to occur.

Ceramic high-impedance devices are different. The use of long lines—say, up to 100 feet—or even the use of high-capacity (cheaper) cable does not attenuate the high frequencies as it does with other high-impedance circuits. Rather, it attenuates the whole frequency range uniformly, unless the insulation happens to be leaky. In the latter case, it attenuates the low frequencies. As ceramic devices have a relatively high output, this attenuation can be suffered, so that long lines do not seriously impair performance. Inexpensive lines of this type are apt to deteriorate with age and use, as well as sometimes being susceptible to microphonicity, so the cable generates a kind of roaring sound in the system any time it is handled while "alive." Also, ceramic devices do not usually have the quality of other types.

Low-impedance lines can feed much further than high-impedance lines (except the ceramic-source connections), up to, say 100 feet. Losses are not frequency-selective, and depend on the gauge of wire used. A heavier wire will enable a longer connection for the same loss. The advantage of line impedance is that it causes minimum losses, and thus can be used for quite long distances, up to thousands of feet. As already mentioned, connections of this type are more expensive. However, the use of line impedance may represent a less expensive solution than installing remote preamplifiers to raise levels so a simpler or less expensive form of connection can be used.

SYSTEM PROVISIONS

On a single amplifier that serves the whole job, you need to be sure that it has all the provisions you need. On a bigger system these will be assembled as separate entities on a console or rack, and the amplifiers will be nothing more than amplifiers, pure and simple. Carefully ascertain all the possible uses to which the system will be put. Will it need to accept inputs from phonograph, radio, tape, etc., in addition to the more obvious microphone for which the initial installation is requested? It costs little more to provide added inputs at higher levels, so see that you have what you need or are likely to need.

It also should have adequate tone control facilities and possibly some form of feedback control to prevent or minimize acoustic feedback. Make sure that adequate controls are provided for what you want to do. Sometimes extra inputs are accommodated by the use of dual controls, so that only one of a pair may be "up" at once, because the control must be turned the opposite way to bring the other input up. This is often quite adequate and it saves having an unnecessarily large number of controls. But if you should want to put on multiple effects, you'll need to have enough controls to work on all inputs independently at the same time. Providing the necessary facilities on the more sophisticated systems is detailed in later chapters, and some of the facilities there described also can be obtained as features on the more expensive single-unit amplifier systems.

Chapter 2

Equalizers, Mixers, and Filters

In all but the simplest installations, equalizers, mixers and filters are needed for one reason or another. In the simplest installations the functions are necessary, but they usually are built into the single-unit amplifier.

USE OF EQUALIZERS

The most universal use for equalizers is in complementary pairs to make the best use of a storage or transmission medium (Fig. 2-1). The ideal system reproduces sound with the various component frequencies in the same proportions received at the input to the system. But these pro-

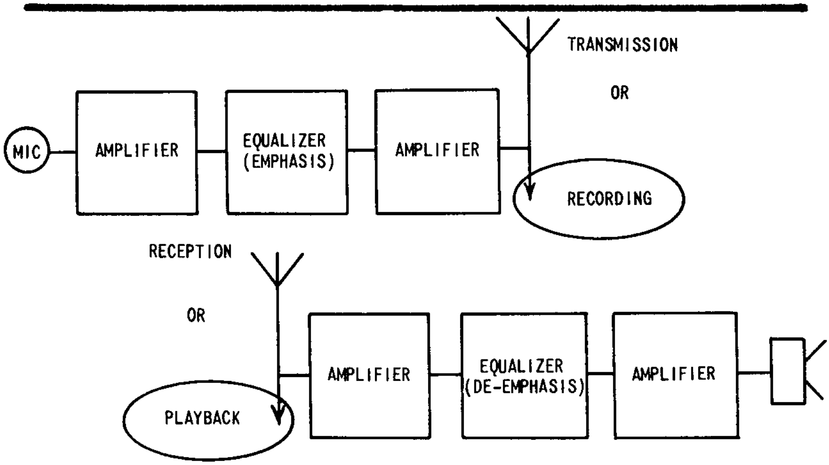


Fig. 2-1. This block diagram illustrates the basic purpose of the most universal form of equalization.

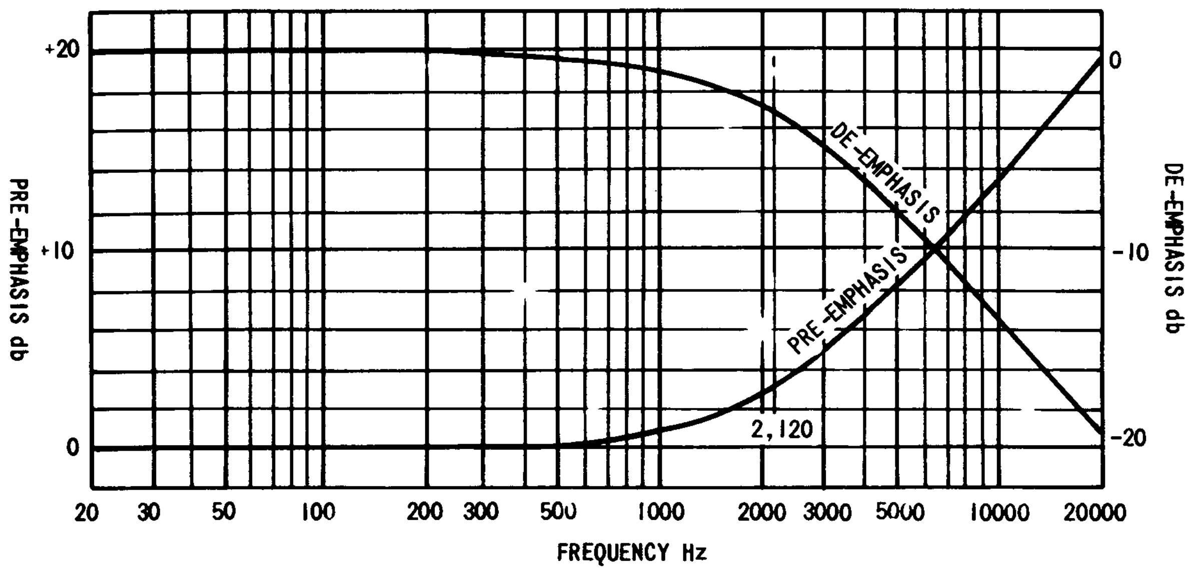


Fig. 2-2. Complementary equalization curves, called pre-emphasis and de-emphasis, used in radio transmission.

portions are not ideal for recording, on disc or tape, or for radio transmission.

The principal factor in all equalization theory is that the higher frequencies in the audio spectrum, while they are the most readily audible, possess the least energy content. Consequently, background noise is most noticeable in the form identified as "hiss." Equalization emphasizes these higher frequencies before recording or transmission, and removes the emphasis during playback or reception reproduction, thus reducing background noise attendant upon the form of record or transmission. So equalization is incorporated in the recording equipment, or at the radio transmitter, to yield a better overall signal-to-noise ratio, and improve dynamic range. Then the complementary equalization is needed on playback or reception to reconvert the component frequencies to their original proportions.

While all equalization serves this general purpose, the precise form varies from medium to medium. For true reproduction, the equalization on playback or reception should complement or match that used in recording or transmission

For radio reception, particularly FM (on AM the aim for high-frequency response usually conflicts with selectivity), the equalization consists of "de-emphasis." In transmission, frequencies above a turnover point of 2,120 Hz are simply emphasized at 6 db octave. The receiver needs to roll off at 6 db/octave with the 3 db point at 2,120 Hz (Fig. 2-2).

For disc recording, European standards differ from the American. So any facility that may handle both sources of records needs to provide equalization facilities for both.

The American equalization for playback, assuming a velocity-type pickup (which most professional pickups are), has three important reference frequencies—50 Hz, 500 Hz, and 2,120 Hz. At the first, the response turns from what would be flat (constant velocity) at frequencies below 50 Hz to a 6 db/octave downward slope (to correct for constant amplitude over this range). At the second frequency the response reverts toward constant velocity (flat) and at the third point it again resumes its downward slope to compensate for constant amplitude over the highest frequencies (Fig. 2-3).

The European or International equalization, while basically

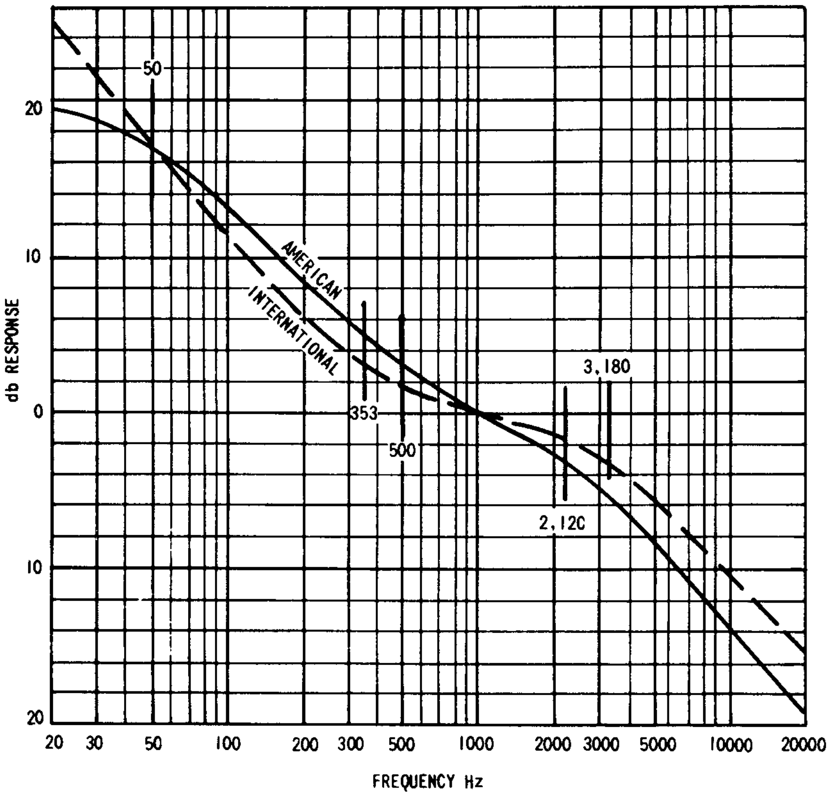


Fig. 2-3. Two playback equalization curves standardized for phonograph records.

similar, uses different reference points. The first turnover of the American curve is missing (constant-amplitude recording extends downward indefinitely from the lower turnover). This, in itself, makes little difference, since not much is recorded that low in the frequency range. Instead of the 500-Hz turnover, the International equalization uses 353 Hz, and instead of 2,120 Hz the International is 3,180 Hz. This means that the overall International playback curve shows more "leveling off" in the middle than the American curve (Fig. 2-3).

A similar difference exists between the American and the International standards for tape equalization. Both use a 6 db/octave downward slope through the major part of the audio range. The American levels off at the low-frequency high-level end, with a turnover point at 50 Hz (like the disc equalization) while the International standard doesn't have this turnover at all. Both level off at the higher-frequency low-level end, the American with a turnover at 3,180 Hz for speeds of 7.5 and 15 inches per second, and at 2,120 Hz for speeds of 1.875 and 3.75 inches per second. The turnover on the International curve is at 1,500 Hz (Fig. 2-4). So any facility that expects to use tapes from all sources needs these different equalizations available.

LOUDNESS COMPENSATION

Another need for equalization arises from playing program at a level different from that at which it is recorded or picked

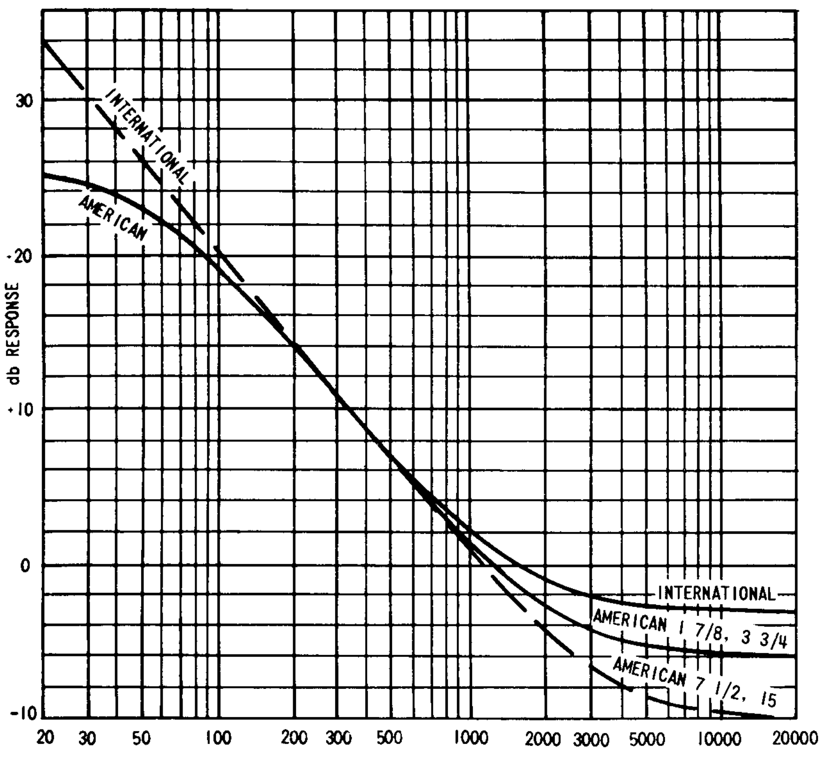


Fig. 2-4. Standard equalization curves for tape recording (curves shown are for playback).

up. For example, background music systems, unless they use specially recorded music, play at a much lower level than the program is normally reproduced. Without equalization it seems to lack character, because all of the lower frequencies, and some of the extreme highs, seem to disappear due to reduced sensitivity of the average human hearing faculty in these frequency ranges at reduced levels.

So programs of this kind need loudness equalization. The same is true for programs containing dialog or action effects with background music. When mixers are used to put such programs together, the channels containing background music need loudness compensation equalization. However, this kind of equalization is best made flexible so it can be adjusted until it "sounds right." For some home high-fidelity systems, the manufacturers incorporated what they termed a "loudness control" in which this compensation was made automatically as volume is turned up and down, which at first seemed like a good idea. A loudness control that incorporates automatic response compensation for changes in setting requires a separate level control for each program input, so the response to each program is correct at all settings of the loudness control. This is needed to compensate for differences in level of recording or transmission that do not reflect intended changes in reproduced loudness.

Some recordings employ a higher peak level than others. In the system these peak levels should all be the same, a limit set by the handling capacity of the system. If the loudness control is used to set levels for this purpose, the equalization at maximum loudness on each program will differ according to the recorded or transmitted volume level of that program. For the professional operator, it proves far simpler to provide separate bass and treble controls. In fact, most high-fidelity systems have reverted to this method, for the same reason.

Tone controls also can be used to enhance the voice quality of individuals before a microphone, especially in an auditorium with poor acoustic qualities, which always seems to emphasize poor speech qualities. One individual may have a deeply resonant bass voice that comes out overly muffled in a poor acoustic environment. Bass needs reducing and maybe treble needs reinforcing to make this individual's voice sound balanced, or even "natural" for that individual.

Another individual may have a shrill, edgy voice, which the acoustic environment also tends to emphasize. The reverse treatment is needed for this person. Just as there may be some argument about whether a camera really presents a true picture, so there is no doubt that a sound system does not accurately represent individual voice qualities but tends to "caricature," if not to outrightly distort, especially in a poor acoustic environment.

Sometimes, rather than accentuating one end or other of the frequency spectrum, a person's voice seems to "grate" on one note, perhaps one that may seem to want to excite an acoustic howl. An experienced speaker will often inflect his voice to avoid the effect, but many who speak into microphones use their normal voice and seem unable to modify it to suit the needs of the moment. So, then, the system should be able to play down that particular area in the frequency spectrum, until that person's voice no longer "grates" at the particular frequency.

Requirements to achieve these variations in different environments will change from installation to installation. Those with better overall acoustics will not require much help in this department, and minimal control, say simple bass and treble lift and cut, may be quite adequate. But a system installed in a poor acoustic environment may need a more comprehensive set of controls, one that divides the spectrum into three or more bands with separate level controls for each. How many such controls are needed also may be subject to variation from one installation to another. In the better acoustic environment, one set of tone controls for the whole system will suffice. In the poorer one it may be advisable to have two or more available for use on different microphone channels. Or, if two people are likely to share one microphone for an interview type presentation, it may be advisable to have alternative controls on the one input, with a key that can change from one to the other, according to which of the parties is speaking at a particular moment.

One more kind of equalizer sometimes used is called a "presence equalizer." It has been found that a reproduced program which sounds "lifeless" or as "lacking presence" can often be enlivened by slightly emphasizing frequencies centered about 3,000 Hz and extending from about 1,500 Hz to about 6,000 Hz. Not more than 6 db maximum boost achieves this, and often a smaller boost is sufficient. An adjustable equalizer of this type is a useful adjunct on some installations.

TYPES OF EQUALIZERS

Whatever equalizers are used, they may be one of two major types—active or passive. The latter are the simplest, but inserting them in a circuit involves a loss of gain. Only at a maximum boost point will the overall gain approach the level of transmission in the absence of the equalizer. This means that additional amplification is needed whenever passive-type equalizers are used, so that normal level is maintained. A 20 db line amplifier is useful for this purpose and consoles with many equalizers usually have an equal number of patchable line amplifiers so the level can be maintained.

Mistakes can be avoided by including a line amplifier with each equalizer, so it is automatically included when the equalizer is patched in. This makes patching more foolproof. But if the two are going to be put together, anyway, why not use the amplifier to improve the equalizer? This leads to the active equalizer, a unit that includes gain so the mid-band level is unchanged from input to output. Feedback is used to reduce distortion to the maximum degree at any frequency. With the active type a boost is really a boost; with the passive type a boost is merely a lack of attenuation present everywhere else.

Example. Fig. 2-5 shows an equalizer stage to give a boost between 50 and 500 Hz. It is calculated as follows:

First, the emitter-follower output stage is voltage-biased for maximum level handling (emitter at half supply voltage). Making the bias resistors each one-tenth of the emitter resistors means that if current gain (beta) is 100 the current in the bias resistors swamps base current by 10 times. The base input resistance will be about 27K (100 times the emitter resistor). The two resistors effectively in parallel (to the signal) total 1.35K, reduced to 1.3K by the parallel effect of base input resistance. Using a collector resistor of 1.2K to the voltage-gain stage makes the total series interstage resistance 2.5K. So the resistance to be bypassed for boost needs to be nine times this to yield a 20 db step for which a 22K resistor serves. Bypass capacitor C1 needs to have a reactance of 22K at 50 Hz (or 2.2K, made up of 2.5K in parallel with 22K, at 500 Hz). This figures to 0.145 mfd. A selected 0.15-mfd capacitor should give the correct response.

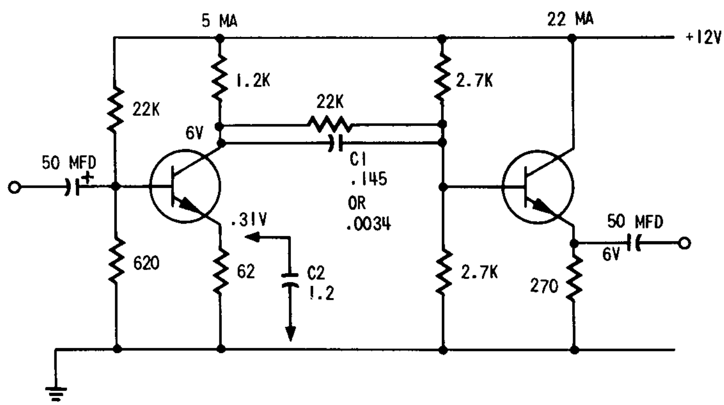


Fig. 2-5. A transistor circuit providing the American equalization for phonograph recordings.

Now to complete the necessary gain. At frequencies below 50 Hz the voltage developed across the collector resistor divides between 22K and 1.3K (input impedance at base), dividing the voltage by 23.3/1.3 = 18. To give the voltage gain stage this amplification, the emitter resistor must be 1/18th of the parallel impedance in the collector, made up of 1.2K in parallel with 23.3K, which is 1.14K. This figures to 63.3 ohms. A 62-ohm resistor will give a slight residual gain below 50 Hz.

As a check on this figuring, the voltage gain should be a little more than 10 when C1 completely bypasses the 22K resistor. Then the collector load is 1.2K in parallel with 1.3K, which makes 625 ohms. With the emitter resistor of 62 ohms, the gain is 10 times its value below 50 Hz. To get the correct operating point for the voltage - gain stage, assume six volts on the collector, which sets the current at 5 ma and requires an emitter voltage of 0.31. Base current will average 50 microamps, assuming current gain (beta) is 100. Making bias potentiometer current 10 times this requires values of 620 ohms and 22K for 0.31v and 11.7v, respectively. This conveniently makes an input load impedance of 600 ohms.

Next, assume we add lift beginning at 2,120 Hz to complete the American recording characteristic. One way to do this would be to boost the gain of the voltage-gain stage beyond the maximum 10:1 already set, starting at 2,120 Hz. This can be done with a bypass capacitor across the 62-ohm emitter resistor. The capacitor needs to have a reactance of 62 ohms at 2,120 Hz, requiring a value of 1.2 mfd. The limit of this lift depends on the associated circuit impedances. The voltage gain is achieved, in effect, by lowering the current acceptance impedance in the base circuit. With a transistor current gain (beta) of 100, the base input resistance is 6.2K.

If the external source resistance is infinite, the base source impedance is about 600 ohms (620 in parallel with 22K). So the top of the lift will come at about 20 kHz. But if the source resistance is 500 ohms, the base source impedance is about 275 ohms and the top of the lift will be at about 10 kHz. If the source is another amplifier whose actual impedance is lower than the nominal 500 - ohm impedance, the lift could be even less.

This method allows the lift to be too dependent on external circuit impedances. Also, as C2 effectively brings the voltage-gain stage to its maximum un-fed-back gain, the natural nonlinearity of the transistor, particularly its base input resistance, will show. This, too, will show up worse when working from a low source resistance. So the better plan would be to add another stage like that in Fig. 2 - 5, but with the lift starting at 2,120 Hz instead of 50 Hz. This can be achieved simply by changing the C1 in the next stage from 0.145 mfd to 0.0034 (a selected 0.0033) mfd. The coupling capacitors at input and output are 50 mfd, which sets the 3 db point for these couplings at about 6 Hz.

The reverse playback characteristic can be applied by putting a similar response in the feedback (Fig. 2-6). As the gain will be controlled by feedback to get the requisite response, the middle stage can use maximum current and voltage gain, which means the emitter is grounded. Signal current input is controlled to one milliamp per volt by the 1K resistor from the input emitter-follower.

The voltage-gain stage is direct-coupled to the emitter-follower output. The emitter of this stage provides stabilized bias for the middle stage, along with signal feedback. If the middle stage has a current gain (beta) of 100, a collector resistor of 1K, a 100K DC feedback to base, it will maintain the middle-stage collector and the output-stage emitter at

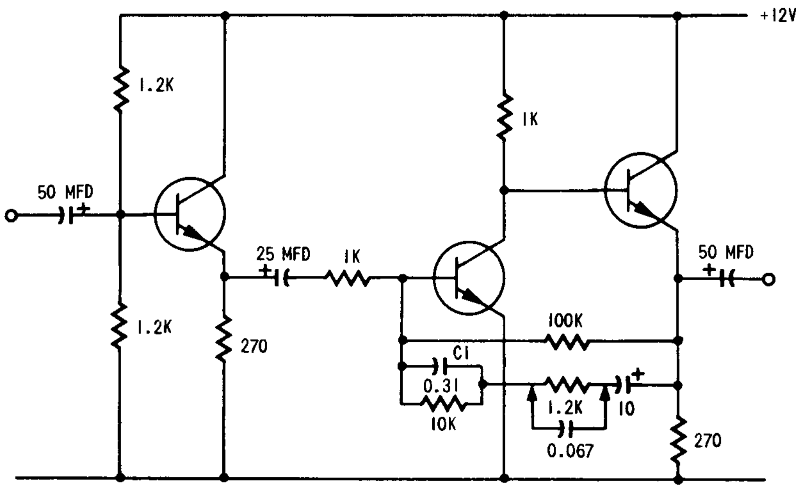


Fig. 2-6. A transistor circuit providing the playback characteristic to complement the circuit of Fig. 2-5.

half the supply voltage. If the gain falls to 70, the collector voltage will rise to 7, and if it rises to 140 the collector voltage will fall to 5. If a simple bias resistor of 200K were used from supply positive, assuming a gain of 100, a change of gain to 70 would raise the voltage to 7.8, and to 140 would lower it to 3.6. Obviously, the feedback produces a worthwhile improvement. Now, a large capacitor couples the signal feedback components. 10 mfd is large enough to avoid introducing any effect on audio response within the audio range.

For frequencies above 500 Hz, the 1.2K resistor will control feedback current so that an output voltage of slightly less than 1.2 times the input voltage from the emitter-follower input stage will balance it, the base current for the middle stage (signal component) being only a small fraction of either. To let the output voltage rise to 10 times as much as frequencies below 50 Hz, the additional resistor, bypassed by C1, needs to be nine times 1.2K to total 10 times. A 10K resistor will serve.

Now, the bypass capacitor should have a reactance of 10K at 50 Hz, requiring 0.31 mfd. If the same circuit is desired to provide a roll-off above 2,120 Hz, another capacitor may shunt the 1.2K feedback resistor, having a reactance of 1.2K at 2,120 Hz, requiring 0.068 mfd. Notice that deleting the 10K resistor shunting C1, which would also dispense with the need for the 10-mfd DC blocking capacitor in the signal feedback path from the output emitter, would result in removing the 50-Hz turnover, yielding the start of the International equalization curve.

To change the next turnover from 500 Hz to 353 Hz, capacitor C1 should have a reactance of 1.2K at 353 Hz, requiring 0.375 mfd. For the high roll-off the value should be 1.2K

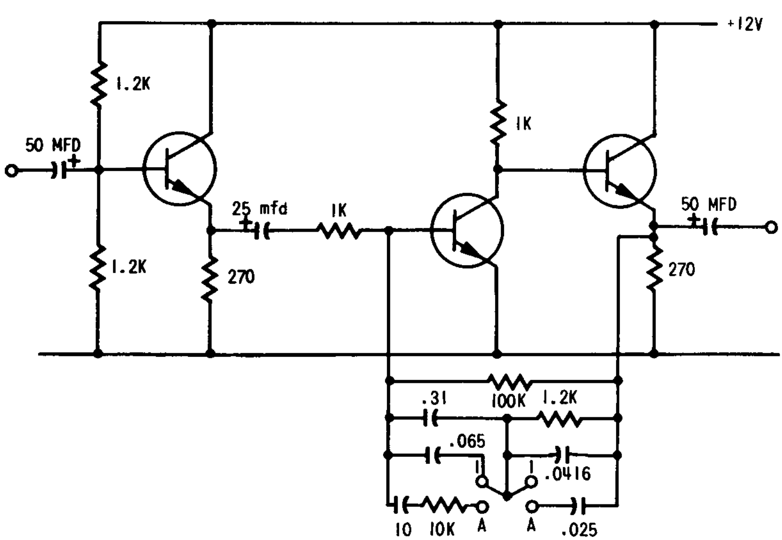


Fig. 2-7. A phonograph playback equalizer with switching to provide American or International equalization.

at 3,180 Hz, requiring 0.416 mfd. Thus, a little rearrangement makes it possible for a double-pole slide switch to change the circuit from American to International equalization. (More on the practical construction of equalizers follows later in this Chapter.)

MIXERS

Mixers are controls whose purpose is to mix various elements of program together—the action center of an audio system. For this reason considerable attention has gone into their design over the years. Two areas have received concentrated attention: (1) making the operation silent and durable; and (2) making it easy to use with an action involving natural human reflexes. Any contact that is used as a "slider," virtually continuously, is liable to deteriorate with long use. This has led to the development of contacts with a minimum tendency to oxidize or wear, and with self-cleaning action.

Making such controls easy to manipulate placed emphasis on knob design so that the position would be easily recognizable at a glance, and also make it easy to change settings, if possible, with only a finger for each control, in such a way that each hand could control several knobs simultaneously. The final improvement that has now become almost universally accepted is a control operated by a linear lever instead of a rotating knob. This not only makes the individual control positions instantly visual but manually functional as well. It also enables closer grouping so that the hand can encompass a greater number of controls for simultaneous manipulation.

Electrically, mixers introduce loss, depending on how many controlled circuits are connected together for mixing. So, as with equalizers, extra gain must be provided so the output is at the proper level. Mixers can work several ways. Most professional mixers of the passive type use T-pads, or if the circuits are balanced, H-pads (Fig. 2-8). Each pad is designed in steps so that movement from one stud to the next changes level by a fixed amount, say 1.5 db, when the pad is correctly terminated, particularly at the output end.

One way of combining mixer pads is to parallel their outputs and terminate the combination with a resistance value 1/nth of their working impedance, where n is the number of circuits mixed (Fig. 2-9). This is the most efficient form of mixing, yielding the minimum loss for a given number of circuits, referred to the full-on position of individual circuits. Its disadvantage, where the sources are passive (resistance), is that loading the output with an impedance much lower than the characteristic value causes the last few steps to act quite drastically; that is, introduce more than the designated db per stud.

For example, if the mixer is a 3-way type, the last few studs give an attenuation interval of 4.5 db per stud, instead of the intended 1.5 db per stud. If the source is active—that is, if each pad feeds from the output stage of a preamplifier or some such circuit designed to deliver a voltage into line impedance—then turning the control all the way up re-

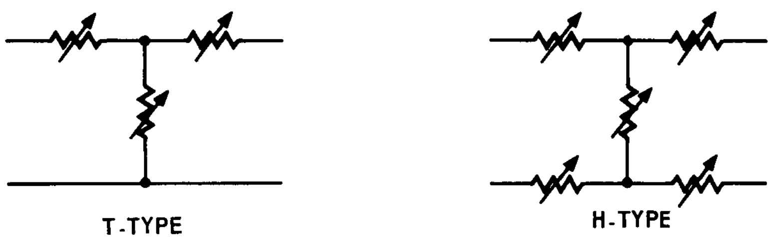


Fig. 2-8. T-type and H-type are the two major attenuator pad configurations used in professional equipment.

sults in loading down the output stage seriously, in addition to this sudden effect at the top end. If more than one circuit is turned all the way up, each is loaded down even more.

Suppose the mixer impedance is 500 ohms and there are six circuits. The terminating resistance will be 83 ohms. The output source impedance, intended to feed into 500 ohms, may be 50 ohms. Turning one control all the way up will cause that control to feed into about 45 ohms, made up of the 83-ohm termination, in parallel with the five other circuits of 500 ohms each, which makes 100 ohms in parallel with the 83. This will make the last few studs shown about six times their rated attenuation, or about 9 db per stud, which is pretty drastic.

But suppose a second circuit is turned all the way up. Now each of the circuits turned up will provide a parallel load for the other—not of 500 ohms—but of 50 ohms. This reduces the loading on each of the two circuits to 83 in parallel with 125 (four at 500) in parallel with 50 (the other source impedance), or about 25 ohms. That last stud will carry about 25 db, which is a considerable level change, quite undesirable. In addition to this loading effect may be the fact that the 50-ohm loading each active source presents to the other may not be a linear resistance, so that the extremely sudden level change also causes distortion, not due to simple overload but due to nonlinear loading.

Before proceeding to a better connection technique, let's compare the result of making the termination the same as the operating impedance of each circuit. This increases the attenuation for a given number of circuits by from 1.3 db to nearly 6 db. In Table 2-1, Column A gives the attenuation caused when n circuits are terminated by a load 1/nth of their individual impedance. Column B gives the attenuation when they are terminated with the same impedance at which each circuit is designed.

Making this change protects against loading an output down when only one circuit is turned all the way up at a time. If more than one circuit is turned fully up, the loading down referred to, of one circuit by another, with the possible distortion it can cause, still happens. The best way to protect against this also prevents some of the unnecessary loss represented by the difference between Columns A and B, but it inserts some additional matching resistors. Fig. 2-10 shows the arrangement. Each circuit is terminated with a series resistor (of value a) so that, when the whole group is terminated with a resistor (of value b) that matches the parallel value of them all, each is correctly terminated at all times.

Table 2-1 also lists values of a and b as decimals of the working impedance of each mixer. Column C gives the insertion loss for the number of circuits used, with the arrangement in Fig. 2-10. And value c is the one used to terminate the arrangement listed in Column A and shown in Fig. 2-9 for comparison.

An alternative mixing method employs each control as a means of varying gain in a single-channel amplifier. This may simplify the circuit of the individual mixer a little. The

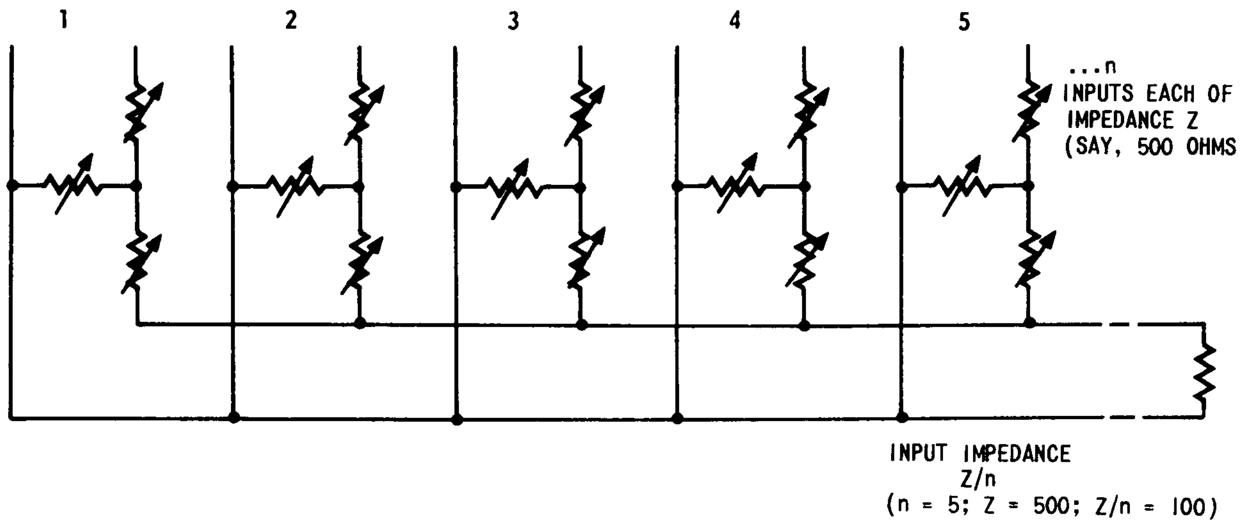


Fig. 2-9. Drawing showing method of combining attenuator pads in a mixer circuit with minimum insertion loss.

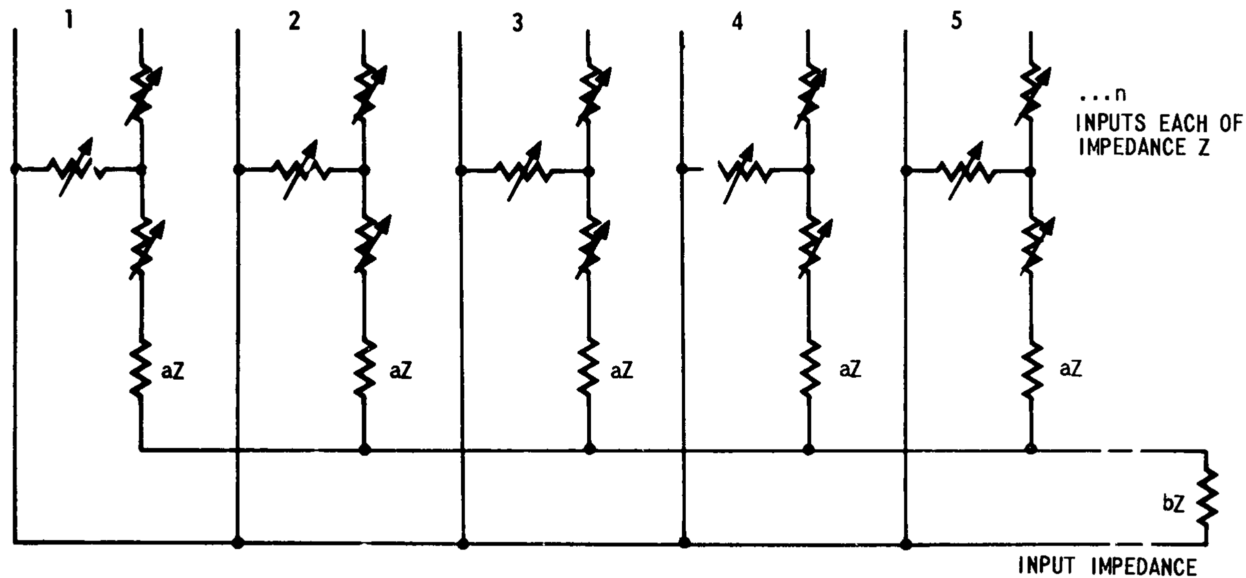


Fig. 2-10. Method of combining attenuator pads in a mixer circuit to provide isolation against interaction that causes excessive last-step attenuation and distortion.

Table 2-1

Attenuation and circuit values for mixer circuits (see text).

Number of Circuits	Attenuation (db)			Values (x Z_0)		
n	A	B	C	a	b	c
2	3.5	4.8	4.8	0.5	0.75	0.5
3	6.2	7.8	7.0	0.67	0.56	0.33
4	7.9	10	8.45	0.75	0.44	0.25
5	9.1	11.75	9.55	0.8	0.36	0.2
6	10.05	13.2	10.4	0.83	0.305	0.167
7	10.8	14.5	11.1	0.86	0.265	0.143
8	11.5	15.55	11.75	0.875	0.234	0.125
9	12.05	16.5	12.3	0.89	0.21	0.111
10	12.55	17.4	12.8	0.9	0.19	0.1
11	13.0	18.2	13.2	0.91	0.173	0.091
12	13.45	18.9	13.6	0.92	0.16	0.083

loss should still be arranged according to the arrangement of Column C, and the circuit shown in Fig. 2-11. Tabulated underneath are the values to be used for different numbers of circuits.

The voltage gain is higher than any in Table 2-1, because this circuit is designed to operate from 500 ohms and into 500 ohms, not a value reduced by the number of circuits. Series resistor c is needed at the output to prevent each circuit from loading all the others and is chosen so that the 500-ohm output load is matched by all the emitter-followers and their series resistors in parallel.

Value b is selected so the gain stage gives the voltage gain listed for that many circuits. Finally, value a is chosen to provide correct bias when value b is used as an emitter resistor. Using 620 ohms from base to ground provides an approximate matching load for the input end. The audio-taper control gives a good mixer range. Yet another form of mixer uses remotely-controlled gain stages. This is described in detail in Chapter 4.

FILTERS

The final component discussed in this Chapter differs from an equalizer in that its object is not to adjust the relative level of different frequency components but to eliminate some of them entirely, or as well as the filter can achieve the separation of unwanted from wanted frequencies. An example of the need for filtering is encountered in FM, but this is not audio. The radio transmission should utilize all the sidebands within the allocated bandspread, and then drop dead, so as not to put spurious sidebands into an adjoining channel.

In AM the filtering can be achieved at an audio frequency. By cutting the audio frequency response off sharply at a frequency corresponding to the maximum sideband width, the modulator is prevented from generating unwanted sidebands. For tape recording there is a different reason for filtering. If any harmonics approaching the bias frequency are allowed, they will produce twitters by heterodyne beat with the bias oscillator frequency. The remedy for this is to ensure that no such frequencies reach the output from the audio amplifier.

A twin-T type filter designed to reject the bias frequency is

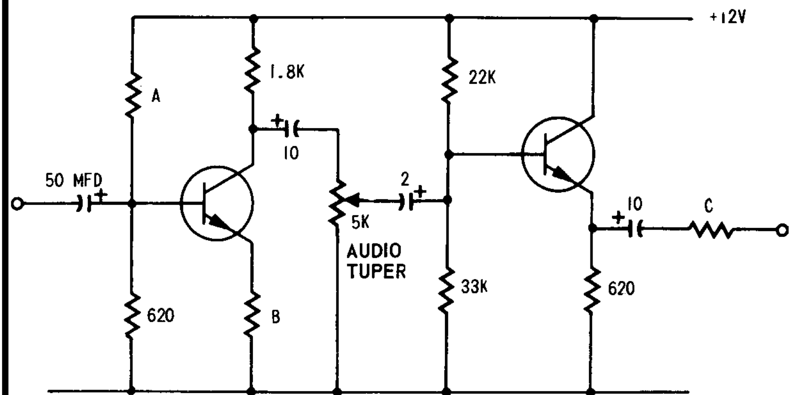


Fig. 2-11. A transistorized mixer element, with values for different numbers of circuits in the total mixer. Input and output impedance = 500-ohms

Number of Circuits	Voltage gain db	Values A	B	C
2	12	7.5K	270	1K
3	15.6	12K	180	1.5K
4	18	15K	150	1.8K
5	20	18K	120	2.7K
6	21.6	21K	100	3.3K
7	23	24K	82	3.6K
8	24	27K	75	4.3K
9	25	30K	68	4.7K
10	26	33K	62	5.1K
11	27	36K	56	5.6K
12	27.6	39K	51	6.2K

one possibility here. The hole is made broad enough, if the frequency it rejects is correctly aligned with the bias oscillator frequency, to ensure that adjacent frequencies are adequately rejected. A filter also may be needed to eliminate a heterodyne whistle from radio reception, or a particularly annoying howl frequency, caused by an acoustic resonance in an auditorium. Or it may be needed to get rid of rumble effects from a phonograph turntable, without cutting the wanted bass frequencies. Also, some of the less well-recorded platters may have rumble recorded on them. While most modern professional quality discs would never have this, one may encounter an amateur recording that needs a rumble filter.

Like equalizers and mixers, filters come in passive and active types, but with filters the reason for needing gain is different. A good filter does not introduce appreciable loss in the pass range, so extra gain isn't needed for that reason, as it is with equalizers.

Earlier audio systems made extensive use of line-derived filters. An m-derived filter could provide a 60 db drop within a very short frequency range. For example, if high-frequency cut-off was to be 15 kHz, the response would be 3 db down at 15 kHz, and 60 db down at 16 kHz. The response "turned off" almost like a switch! But passive filters invariably use inductors as well as capacitors. Unless the filter is operated at a fairly high level, in which case the inductors may cause distortion, they are susceptible to inductive pickup of hum or other stray fields. And such extremely sharp transitions from pass to reject are seldom vitally necessary.

The passive filter is designed around an iterative impedance, which makes termination very important. Wrong termination can seriously invalidate its performance. For these reasons, and to keep distortion low, active type filters are preferred these days. The active filter usually employs active elements to provide effective isolation from terminating circuits and to avoid the other disadvantages of the passive type.

One form of passive filter that doesn't require matched termination is the twin-T, used for rejecting certain frequencies. The transmission of the network may have a null at the frequency of a heterodyne whistle to be eliminated, or at the frequency of an acoustic howl that gives trouble. Other variations possible with the twin-T are that it can produce either a perfect null, resulting in complete elimination of that frequency, or with a different set of values it can produce a dip in the response. This useful circuit employs no inductors, can be built with various "sharpness," but it is best contained between amplifying or at least isolating stages, because it needs to operate from a low source impedance into virtually an open circuit (or vice versa) for optimum separation.

Feedback also can sharpen the separation achieved by a twin-T circuit. Naturally, it makes a fairly wide "hole" in the response (Fig. 2-12). Applying overall feedback sharpens response by delaying the level at which it goes into action. An advantage of the twin-T is that it produces complete

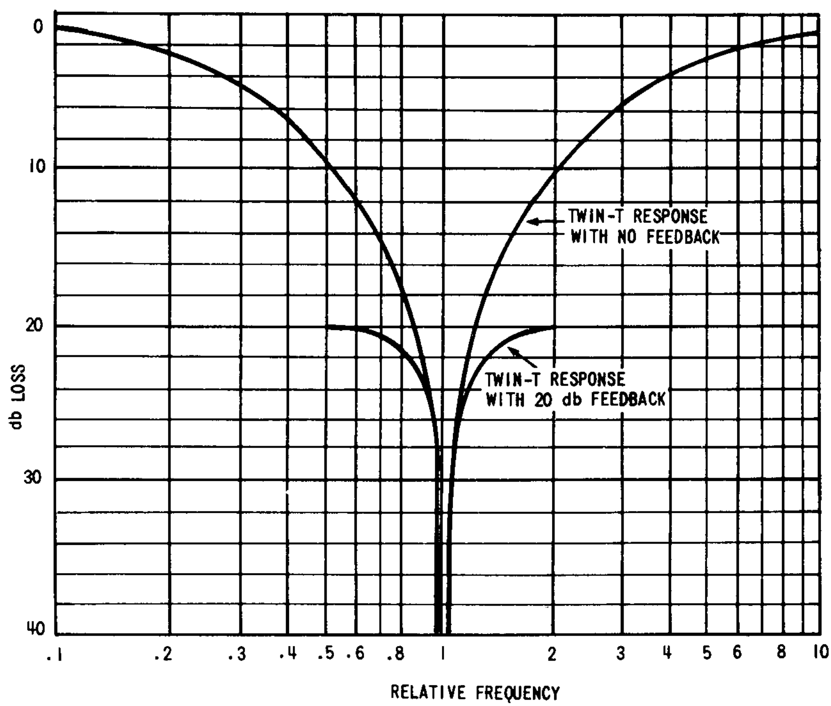


Fig. 2-12. Curves illustrating how feedback can sharpen up the rejection of a twin-T network.

elimination of the unwanted frequency when the values are correctly chosen, which an LC rejection filter can do only by incorporating it in a bridge network.

Most other filters take the form of high-pass or low-pass functions, with an adequately sharp transition from pass to reject. Active types are usually designed on the basis of so many db/octave cut-off in multiples of six. At the cut-off frequency the loss is 3 db and the slope is half the ultimate named for the configuration. Thus, a 24 db/octave filter has a loss of 3 db at cut-off and a slope of 12 db/octave at that frequency. An octave beyond, the attenuation is 24 db, increasing at 24 db/octave (Fig. 2-13).

Adding simple RC combinations in cascade increases attenuation in the desired direction, but the effect is very gradual (Fig. 2-14). However many sections are used, with-

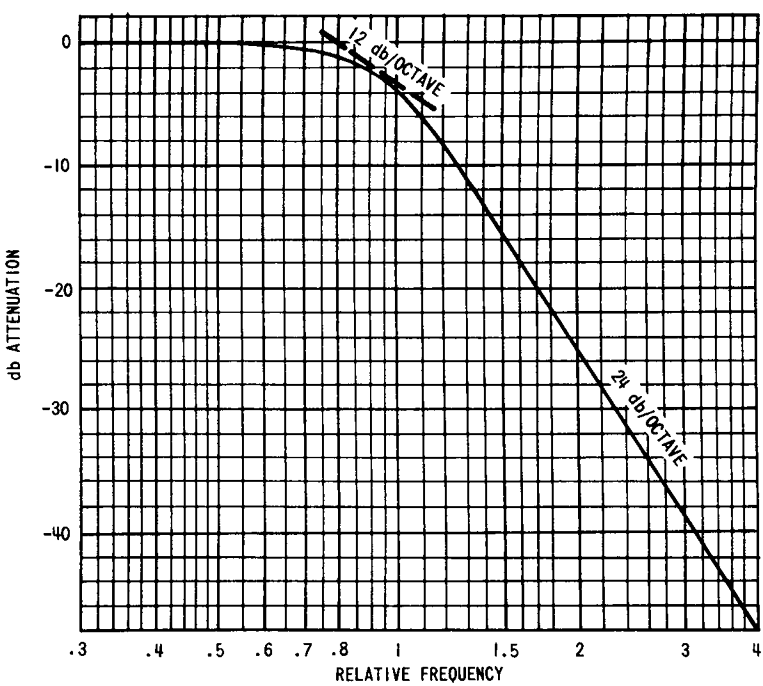


Fig. 2-13. Basic features of a low-pass filter response curve with a 24 db/octave cut-off slope.

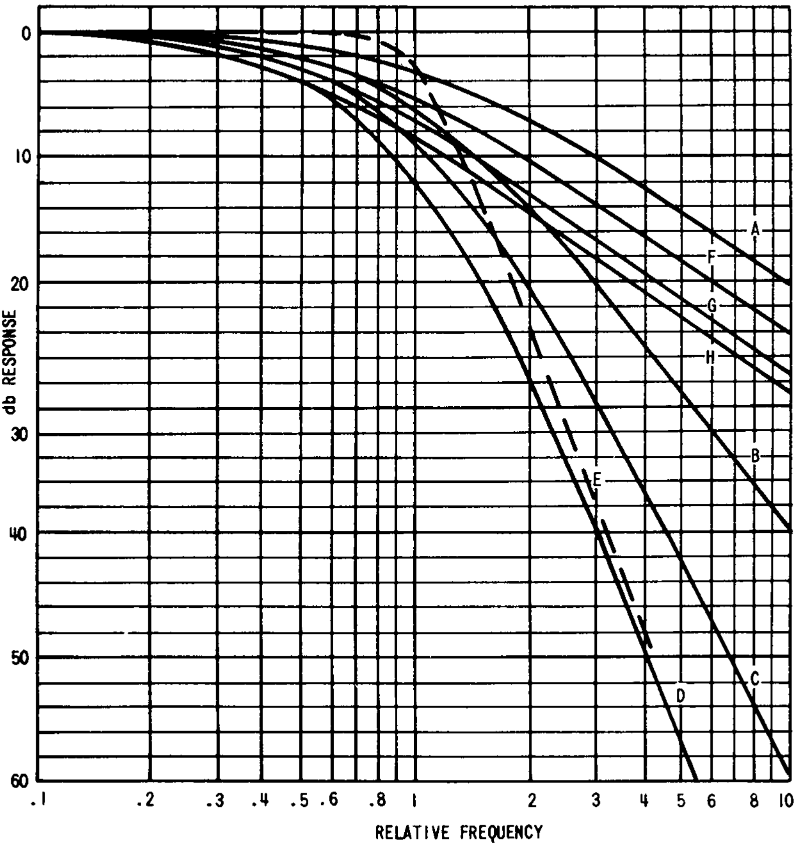


Fig. 2-14. This family of curves points out how multiple roll-offs fail to provide the correct increase in cut-off: (A) single roll-off at reference frequency; (B) two roll-offs at the same frequency; (C) 3 roll-offs at the same frequency; (D) 4 roll-offs at the same frequency; (E) the filter response desired (dashed line); (F) a single roll-off that gives sensibly the same response as (B), up to its 3 db point; (G) a single roll-off sensibly the same as (C), up to its 3 db point; (H) a single roll-off sensibly the same as (D), up to its 3 db point.

out using active elements to modify the result by means of feedback, the response at the end of the pass range (up to 3 db loss) will not differ significantly from that of a single RC combination. The ultimate slope becomes steeper by adding more sections, but the attenuation at the mid-phase point increases 3 db per section. Moving back to the point where the loss is still only 3 db, the performance from the pass range to this point is not significantly different from a single RC roll-off. The increase in roll-off rate occurs beyond the 3 db loss point.

The function of feedback to sharpen the turnover is similar to that illustrated for the twin-T network. But its application is not so simple to apply as in the case of the twin-T. A twin-T network never causes more than 90° phase shift, unless it produces phase reversal instead of null at the critical frequency. Thus, the usual twin-T network can never become unstable, and the addition of feedback without further phase shift (in addition to the twin-T phase shift) cannot increase gain at any point; it can only reduce it.

But when multiple RC combinations are used, the phase shift may rotate much more than 90°. For example, if four identical roll-offs are combined, the total loss at the cut-off frequency will be 12 db and the phase shift at this point will be 180°. What would be negative feedback in the pass range becomes directly positive at this point. In the pass range, where feedback is negative, feeding back a signal equal to original input without feedback results in 6 db feedback. But feeding back this much at the 180° point in this network would result in oscillation at that frequency.

Applying 6 db feedback in the pass range, which reduces gain by that amount, will reduce loss at the 180° point to about 3.5 db, which is only 0.5 db more than the 3 db required at this point for correct filter response. Although pass-range gain is cut by 6 db, the gain is actually increased by feedback at the 180° point by 2.5 db. And at a frequency approximately 5/8 of this (inside the pass range) there is about 2.3 db boost (Fig. 2-15). So adding feedback to sharpen response in the required way is not a simple matter.

An understanding of the procedure can best be attained by starting with a two-roll-off arrangement; in fact, this is the easiest to use as a basic filter element, however sharp your ultimate cut-off wants to be. If two roll-offs are used, separated by a gain stage, and feedback is applied overall, 6 db of feedback extends the turnover point by half an octave, the 3 db point is up 3 db, from 6 db down to 3 db down.

In effect, feedback pushes the 6 db/octave slope, 90° phase-shift point down a 6 db/octave "touch line" (dashed in Fig.

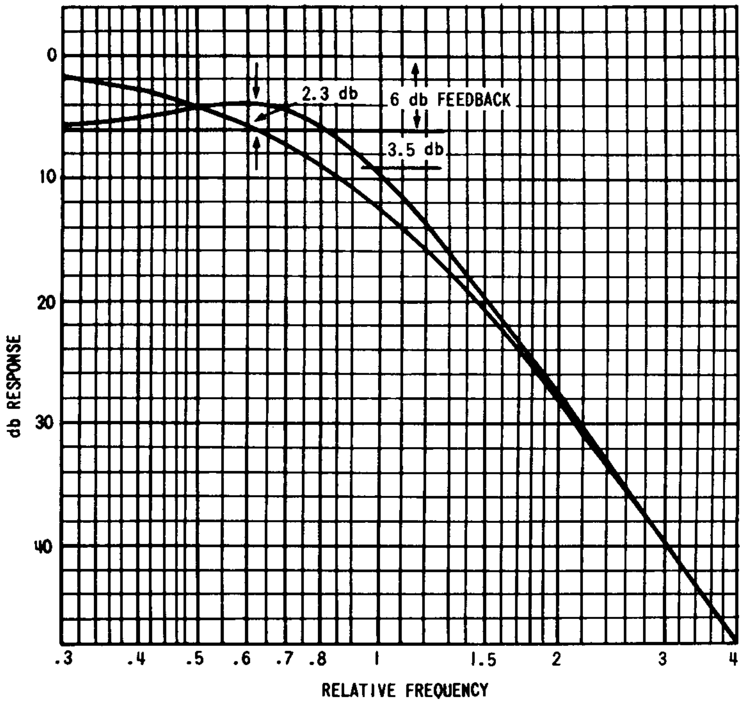


Fig. 2-15. These curves show what happens when feedback is applied over four roll-offs (curve D, Fig. 2-14).

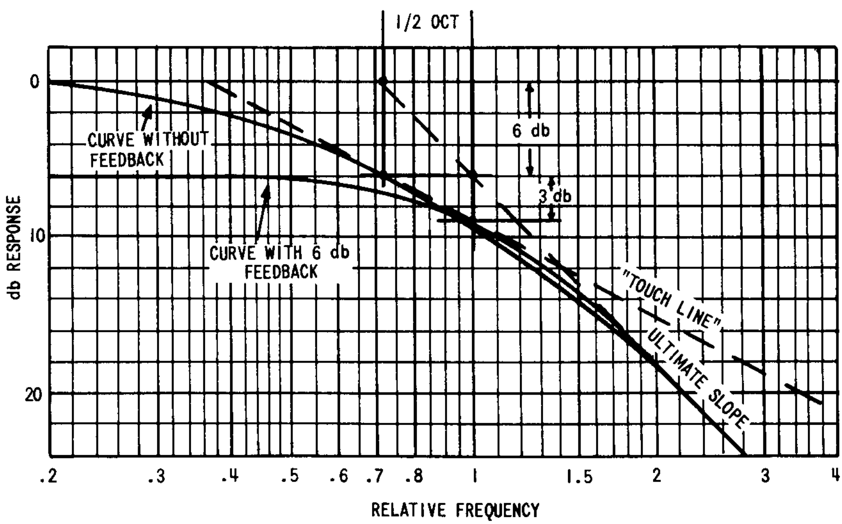


Fig. 2-16. This is how feedback builds the 12 db/octave filter response.

2-16), while the ultimate 12 db/octave slope does not shift. So, relative to the new mid-band level, the 3 db point is up 3 db, from 6 db down to 3 db down. That gives the 12 db/octave filter, for which basic configurations are shown at Fig. 2-17, for low-pass and high-pass. The slope at the 3 db, 90° phase-shift point is 6 db/octave.

The 24 db/octave response is relatively simple to get, merely by adding two roll-offs outside the feedback loop and increasing the amount of feedback used. Fig. 2-18 shows the relevant parameters for both low-pass and high-pass filters of this type, and Fig. 2-19 shows how the response builds up for a low-pass case.

Without going into all the design method, one thing necessary is to guard against feed forward in these networks. Calculations for feedback are based on the notion that signal feeds from the output end of the feedback resistor to the input end. But a resistor has no means of controlling the direction in which signal flows. It will flow from the end where level is greatest toward the end where it is least. The resistor provides feedback as long as the signal at the output end is greater than that at the input end. But as the reactances reduce transmission, there comes a point where the signal at the input end is greater than that at the output end. Then, the feedback resistor can become a "feedforward" resistor, bypass-

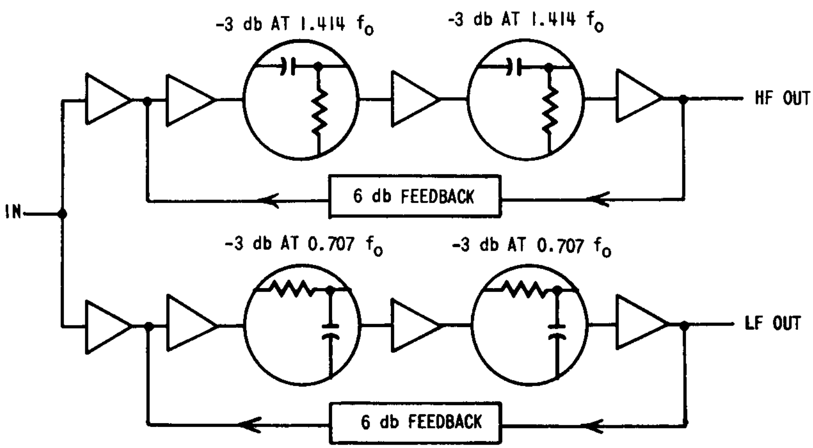


Fig. 2-17. Basic circuit buildup for a 12 db/octave crossover.

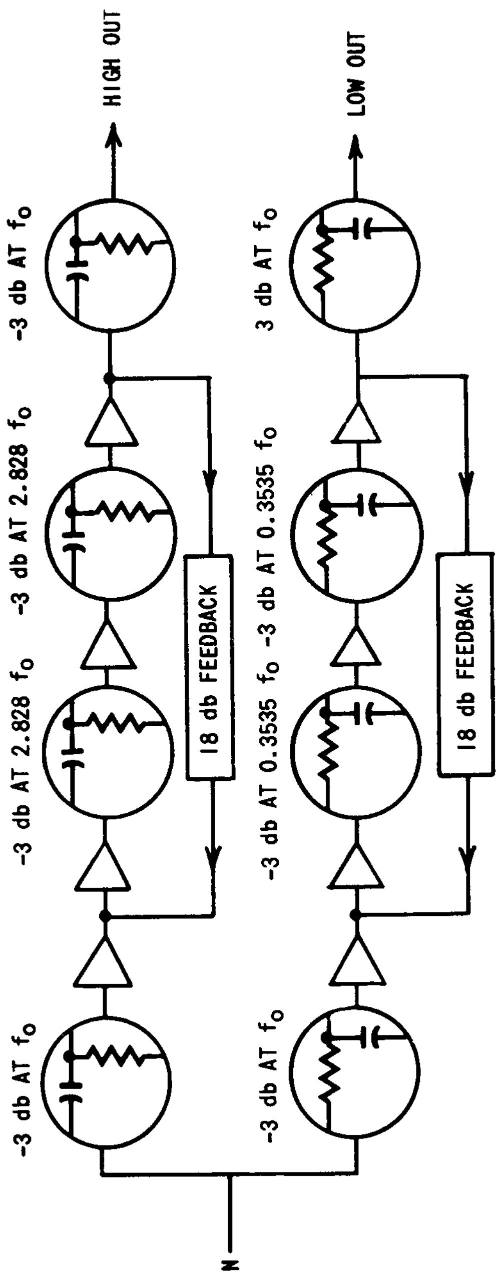


Fig. 2-18. Basic circuit buildup for a 24 db/octave crossover.

ing the amplifier with the reactances that are causing cut-off. This limits the depth to which response will go (Fig. 2-20).

Fig. 2-21 shows a circuit that provides a 24 db/octave crossover with balanced input and outputs. The input balance is provided by a dummy half load, connected to the lower terminal, to match the active half, connected to the upper terminal, which starts the frequency splitting. The output balance is achieved by using phase-splitter stages.

Interaction with the external roll-offs, which use capacitors

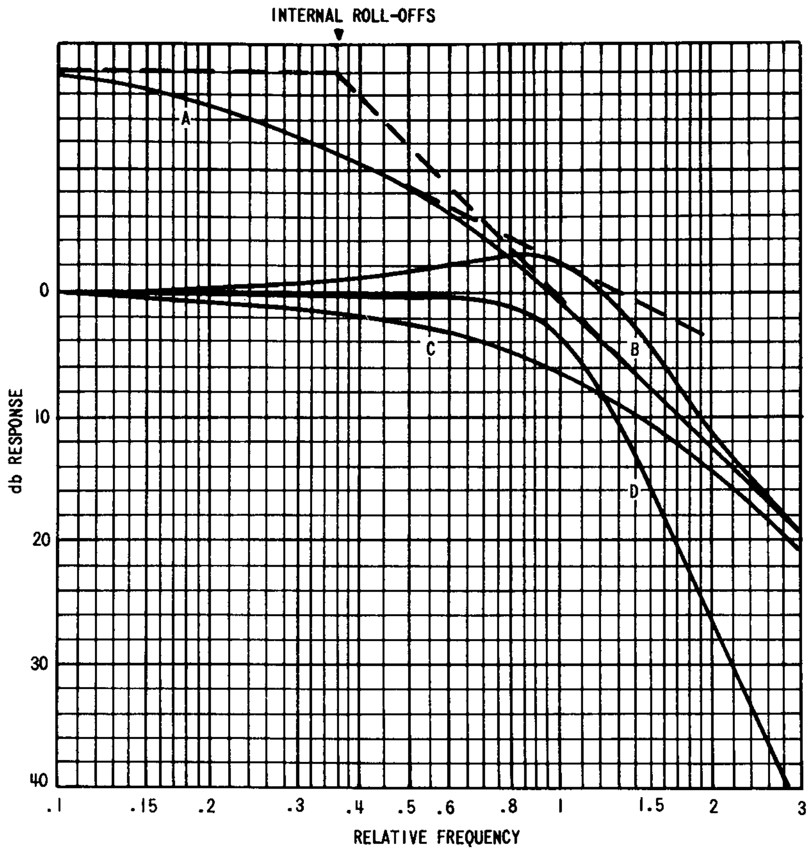


Fig. 2-19. Response curve development of the low-pass part of Fig. 2-18: (A) response of two internal roll-offs without feedback applied; (B) response of internal roll-offs with 18 db feedback applied; (C) response of two external roll-offs; (D) completed 24 db/octave cut-off, adding curves B and C.

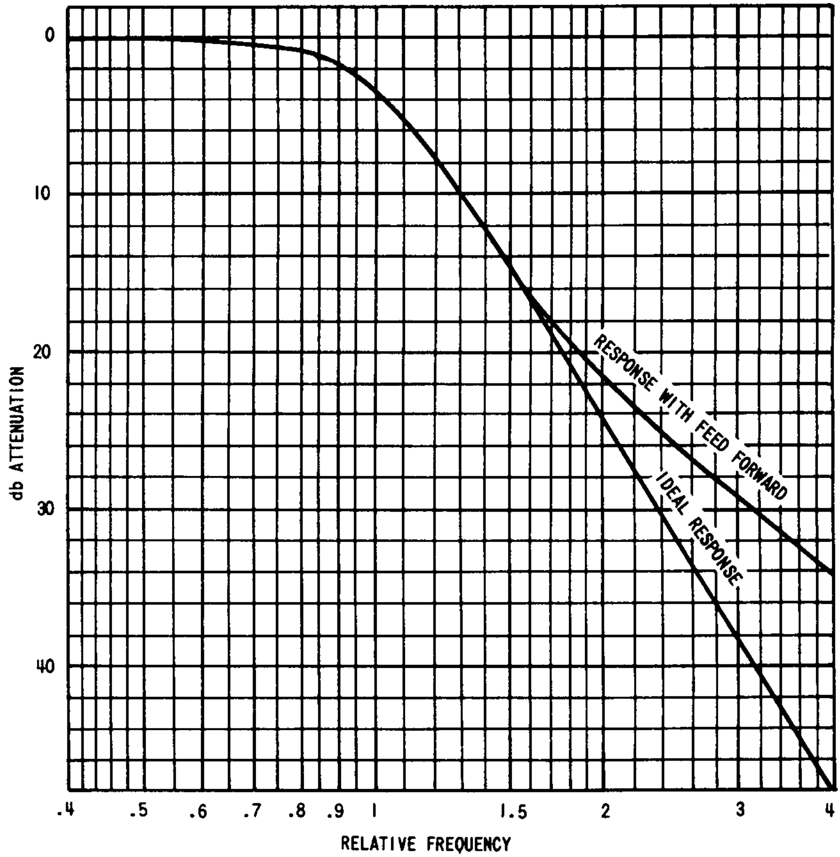


Fig. 2-20. Curves demonstrating how "feedforward" invalidates electronic crossover performance.

in the input and output circuits, is prevented by having enough gain overall to allow attenuation to separate these roll-offs from the internal parameters, which use capacitors between stages. This also enables the turnover points to be sensibly independent of input and output impedance termination.

Individual stage gain is controlled so that the roll-off in each interstage coupling does not reflect any impedance into the stage before or after. Capacitor values are calculated with reference to both the source and load impedance seen by each. And overall gain is calculated to just offset all the losses in the pass range, when terminated with nominal values. All

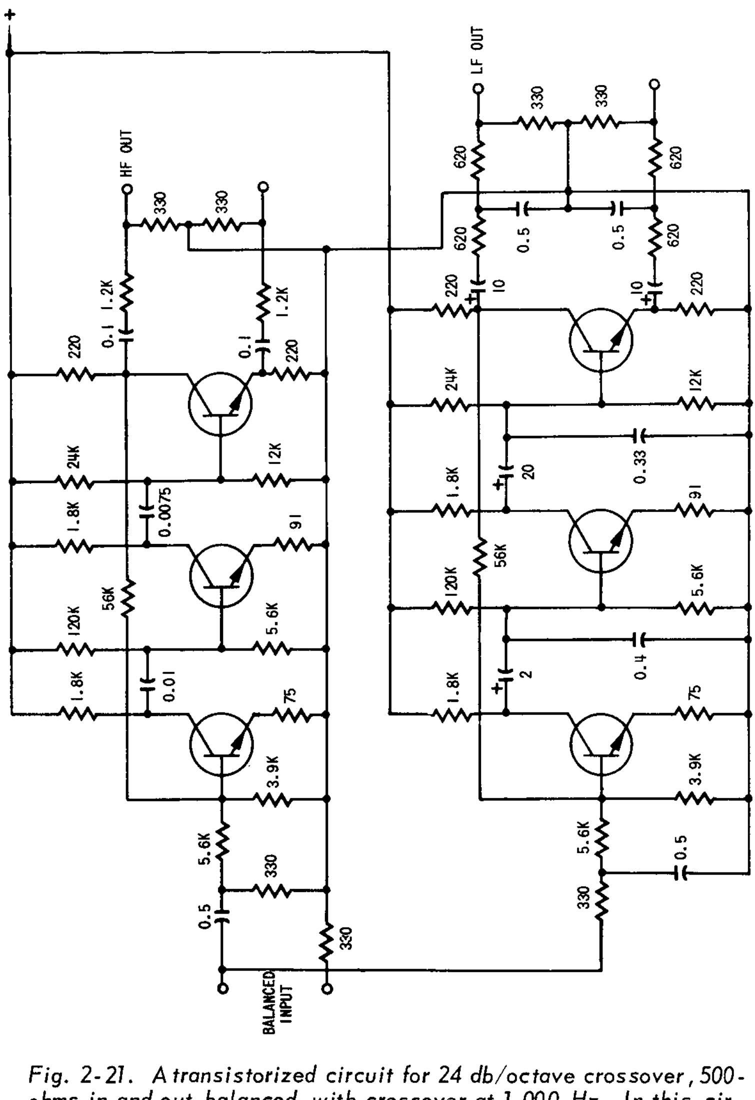


Fig. 2-21. A transistorized circuit for 24 db/octave crossover, 500-ohms in and out, balanced, with crossover at 1,000 Hz. In this circuit feedforward begins to take effect 6½ octaves from crossover, at a point where the attenuation is approximately 156 db.

values are calculated for a crossover frequency of 1,000 Hz. The coupling capacitors in the low-pass unit are made large enough to pass the low-frequency end of the range. The inter-stage values use a staggered roll-off, because 18 db feed-back is enough to cause peaking, as it does for the cross-over, but external roll-offs will not offset it at the low-frequency end of the range.

All transistors are assumed to have a beta of 100. The biasing and gain stabilizing arrangements are such that performance will not be seriously invalidated by gain variation over a range from beta = 70 to beta = 140. The active filter approach is particularly useful for rumble filtering, because a passive filter for this involves large-value inductors and capacitors, which make elimination of hum from the filter quite difficult. A filter that eliminates rumble from the signal, but picks up hum in its stead, is of little use. More sophisticated devices, for purposes similar to those for which filters and equalizers are employed, are described in Chapter 5.

Chapter 3

Distribution Systems

The physical aspects of a distribution system involve the acoustics of the building in which the installation is made. The system has to overcome difficult conditions, such as high reverberation, a tendency toward acoustic howl, avoiding direct feedback to microphone locations, overcoming high sound-level backgrounds, and so forth. Determining what a distribution system must do to achieve its objectives in these directions, for a particular installation, is not within the scope of this book. But the ways and means of doing it, which involve the electronic parts required to do the job, are dealt with in this Chapter.

SPEAKER IMPEDANCE CHARACTERISTIC

One thing all loudspeakers have in common, which affects the preciseness (or lack of it) of all discussions about power distribution, is an impedance characteristic that is far from resembling a constant resistance. The only speakers in common use, after years of varied developments within that type, are moving coil, also called dynamic. The motion of the diaphragm has a direct bearing on the electrical impedance. If the voice coil were locked so it could not move, and thus drive the diaphragm, a loudspeaker's impedance would be that of a resistance combined with inductance (Fig. 3-1).

As soon as the diaphragm is released so the voice coil can move it, this movement reflects definite impedance components due to that motion, the most marked is the rising peak due to the speaker's resonance. Some of this shows in the acoustic response (not shown in Fig. 3-1) where sound output is measured against a constant input to the amplifier driving the speaker as frequency is slowly changed. But the fact that a resonance point occurs indicates the device is inefficient—the air column by which sound radiates fails to load the movement so as to prevent the resonance, and the principal effect reflected into the electrical circuit is due to the mechanical mass and compliance of the moving system, rather than to the air column it moves to create sound. Between the top two curves on Fig. 3-1 the part of the impedance representing radiated sound is shaded.

Because ordinary cone speakers are so inefficient, when compared to compression horn driver speakers, the resonance is not so prominent (Fig. 3-2). In these units the air column is more closely matched to the diaphragm driving it, acoustically, and thus a greater proportion of the impedance measured over the active range of the horn is due to radiated sound energy.

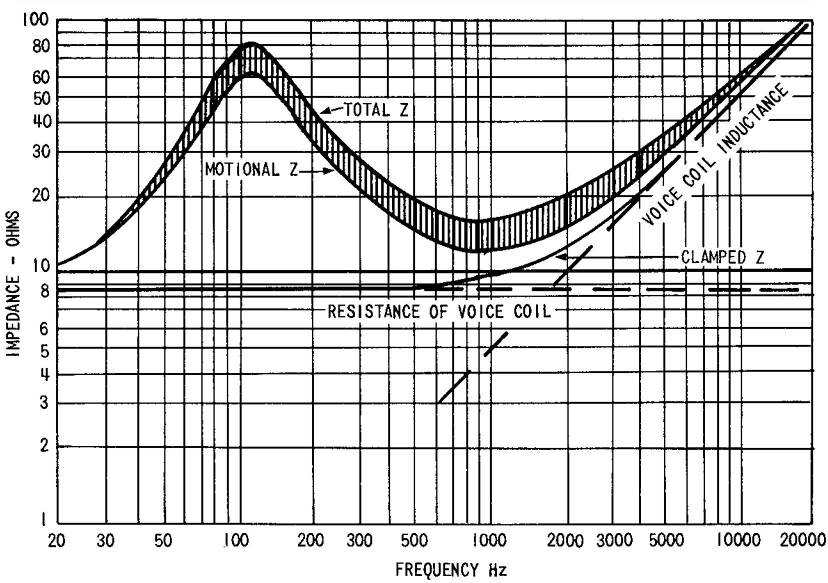


Fig. 3-1. Impedance characteristic analysis for a typical moving-coil (cone-type) speaker. The purely electrical part, measured with the voice-coil clamped, consists of resistance and inductance. The major part of the reflected impedance is mechanical, due to mechanical motion of the assembly. The shaded area is due to acoustic radiation as sound.

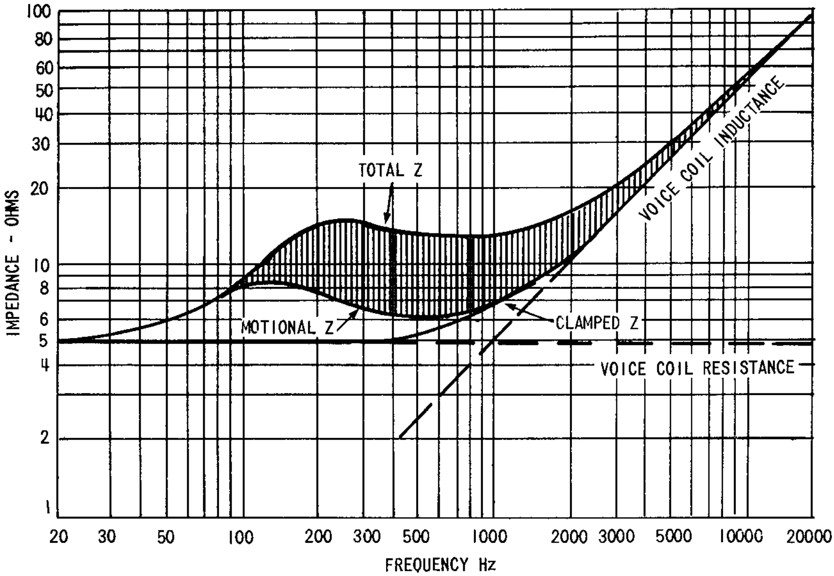


Fig. 3-2. The same graphic representation for a typical compression driver unit.

EFFICIENCY

As was discussed in Chapter 1, a problem of realizing all the power provided by an amplifier arises from the variation in load impedance due to the speakers connected to it. All this means that advantage comes in two ways from using speakers of higher efficiency. Because their impedance is subject to less variation over the frequency range, more of the amplifier power can be consistently used by the speakers connected to it. And because the speakers are more efficient, more of this power is converted into sound.

Some have advocated the use of low-efficiency speakers, on the basis of quality and compactness. In high fidelity systems the so-called "bookcase" or "bookshelf" speaker is highly inefficient, but when sufficient power is provided from an amplifier to drive it, a high quality sound results. But for the same reasons that high-efficiency units improve power utilization, using low-efficiency speakers imposes a much greater demand on available power to achieve equivalent performance. When a 50-watt amplifier is used to drive one low-efficiency speaker, the 50-watt rating means that the amplifier is capable of feeding a 50-watt resistance load. The impedance variation of the low-efficiency speaker probably means that the amplifier seldom delivers more than 10 watts peak, or probably 1 or 2 watts average in a loud passage from a musical combo; less than that from a full orchestra at crescendo.

But, before you consider the efficiency of the speaker, the efficiency of electrical coupling must be brought into perspective, since it may be regarded as of the order of 2%! Therefore, if the speaker is from 2 to 5% efficient, which is typical for this type, the overall efficiency may be of the order of 0.1% or lower! Obviously, using a compression driver system, having an efficiency of between 25 and 50%, with the amplifier coupling efficiency in the same order, will raise the overall efficiency to somewhere between 6 and 25%, which is from 60 to 250 times the available acoustic output using low-efficiency speakers on the same input power.

As any large system faces a dynamic range problem, because maximum power cannot greatly override maximum background noise, due to crowds or extraneous noise, the use of high-efficiency speakers with high-power amplifiers is becoming ever more essential. With these simple facts about efficiency for an audio system, we will proceed to consider distribution systems in a little more detail. First we deal with the simple distribution system intended to deliver the correct power level to a number of loudspeakers, located around the installation.

CONSTANT-VOLTAGE LINE SYSTEMS

Back in the days of tube amplifiers the "constant-voltage line" system was devised, using amplifiers that provide one or more outputs, based on the provision of a specified maximum voltage, to correspond with maximum output.

The term "constant" caused some confusion when that system was first introduced, until it was understood that the word applies to maintaining the same output voltage independent of the number of speakers connected, and not to maintaining the actual voltage constant at all times. Naturally, both output power and voltage fluctuate with signal. But, assuming an instant of maximum signal, the output voltage is constant, regardless of whether the system is feeding one or a hundred speakers (provided it's working within its capacity).

A difficulty in understanding like this occurs only when a system is new. As it comes into everyday use, the people whose job it is to apply the formulas never think about the word used to describe it any more. That it works is all they care, for the most part, anyway. So the constant-voltage system uses a transformer at the amplifier output to provide the so-called constant voltage, like a power line, except that on an audio system neither voltage nor frequency content is constant. Then, too, each speaker uses another transformer, designed to take the designated power (meaning the maximum level) required for that loudspeaker.

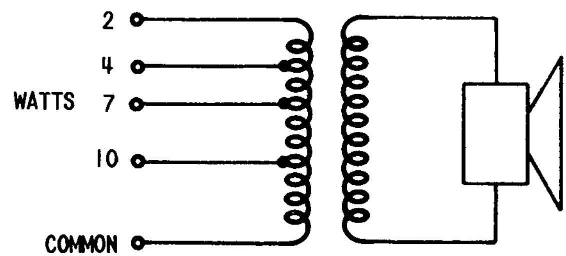


Fig. 3-3. Diagram of a speaker line transformer designed for constant-voltage distribution. Primary taps provide for different power allocation to this individual speaker.

In the constant-voltage system the power level at an individual speaker may be varied by changing taps on its transformer. Such transformers are available with a range of taps—0.75, 1.5, 3, and 5 watts; or 2, 4, 7, and 10 watts (Fig. 3-3). Within an area to be served with sound, the relative level of individual speakers may be adjusted to obtain uniform overall coverage simply by changing taps on the individual speaker transformers. But sometimes changing the relative level in larger area units, involving several speakers in each, is needed. This can be achieved in one of two ways. Amplifier output transformers may provide more than one output voltage. The level of one area relative to another can then be switched by changing amplifier output taps (Fig. 3-4). Such output taps are usually balanced to aid stability. Unbalanced outputs at the voltages used for constant-voltage lines would tend to feed back into input circuits too readily, making it difficult to avoid electrical instability.

However, another approach is more commonly used to provide variable level in large blocks involving a number of speakers in each. In the interests of standby (having equipment in reserve for the event of failure) power output amplifiers should serve only a section of a large system. If one big amplifier is used to serve an entire large system, standby requires a duplicate also capable of serving the whole system, which just doubles the cost of the power amplifier part of the installation. By using a number of smaller amplifiers, each to serve a part of the load (the likelihood of all the small amplifiers failing simultaneously being extremely remote with modern high reliability design), one standby amplifier can be held in reserve for a number of amplifiers in operation. Thus, the additional cost for standby becomes fractional. Also, using different power amplifiers to feed different sections provides a bonus for controlling levels, since the level at each power amplifier can be varied to suit a particular situation. Thus, a whole section of the master control panel may be devoted to controlling the level in different sections of an installation.

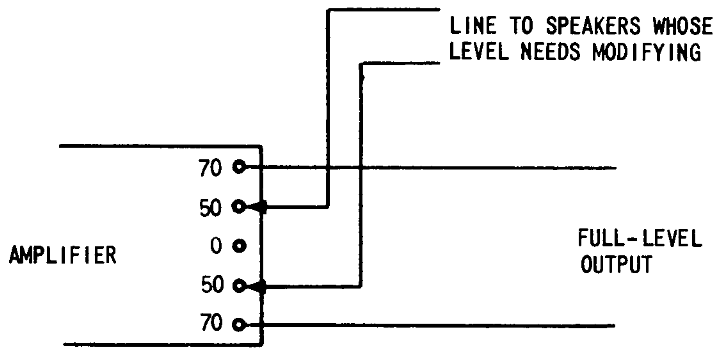


Fig. 3-4. Multiple taps on the amplifier output enable a change in the relative level of blocks of speakers.

In sections of an installation where microphones may be used, such as at a reviewing stand in a stadium installation, loudspeaker levels should be controlled separately from the rest. Then, when those mikes come into use the level of the nearby speakers is turned down, relative to those in the remainder of the installation. A useful method is to provide duplicate output level controls for that section. When the mike is keyed on, extra contacts on the same switch (but very well separated and shielded) switch from one output level control to the other (Fig. 3-5).

One control sets the level to be compatible with the whole system when the mike is not being used. The other sets level a little lower to avoid feedback when the mike is on, but still allows any patrons in that section to hear by listening a little harder. This convenience makes it possible to set all levels under working conditions while still providing for microphone facilities at strategic points. Levels can be adjusted as

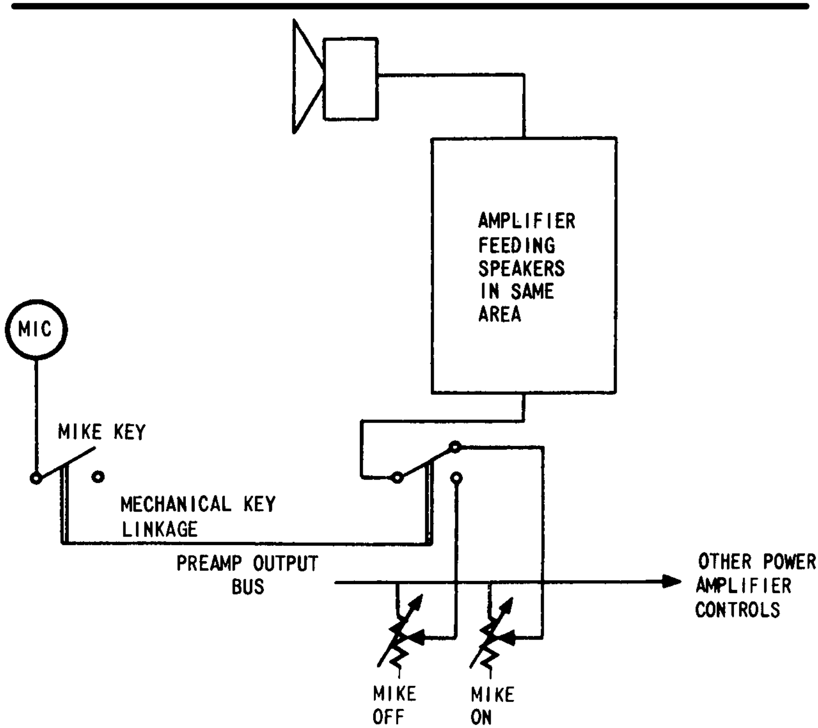


Fig. 3-5. This switching arrangement allows control of a local speaker group when a microphone in the service area of that group is in use.

needed, for example, by a change in audience occupancy, but the general level will be close when the key is first thrown.

Constant-voltage distribution offers a useful standard in making connections. It simplifies wiring, at least between all the units whose level is to be controlled by the same knob at the console. Obviously, a separate line must be brought back for each independently-controlled section. Constant-voltage distribution may be regarded as equivalent to separate fusing of an electrical power installation. The main difference between the two systems is this: fuses don't usually go out unless something is wrong, and they never vary the voltage they pass; whereas, sections of an audio system may be turned either off or down. But the wiring concept is similar.

A constant-voltage system is by no means foolproof. The totalled product of the number of speakers connected, each multiplied by its individual rating (as connected at its respective transformer, not the nominal power handling capacity) must make up not more than the power available from the amplifier to which they are connected, or the amplifier will overload, failing to give its rated output before it distorts seriously, and delivering considerably short of the rated power to each speaker. It's something like figuring the fusing of a power installation. If you put too much on one fuse, it will blow. If you put too many speakers on one amplifier output, that amplifier will overload.

An advantage originally claimed for the constant-voltage system, as opposed to direct wiring at voice coil impedances, was the saving in losses. For example, a system of 20 speakers each with 16 ohms impedance, would provide a load of 0.8 ohms. If the majority of these units were located at the end of an appreciable length of cable, the cable might consume as much audio power as the speakers, unless the cable is of quite heavy gauge. Using the higher voltage and associated impedance avoided this loss, or reduced cable cost.

In those days, an alternative was series connection. Connecting 20 speakers of 16 ohms in series makes a load of 320 ohms, which would eliminate transmission losses. But then the reliability factor came into question. In those days, voice coils more often went open than short-circuit. So one failure would disconnect the whole system, and it would take some time to find out which one had "gone." Of course, series-paralleling would effect a compromise. By connecting groups of four speakers in parallel, and then connecting the five groups in series, the impedance of the system would be 20 ohms (Fig. 3-6). But this approach involves some figuring to find the best way of doing a particular installation. The constant-voltage approach seemed so much more flexible, and involved no figuring except simple addition of wattage ratings.

But the constant-voltage approach was born when tube amplifiers were in vogue and transistors had not been invented. In a tube amplifier you virtually had to use an output transformer to isolate the high-voltage B-plus, and to match the distribution impedance (whatever it was) to the plate circuits of the output tubes. As an output transformer was nec-

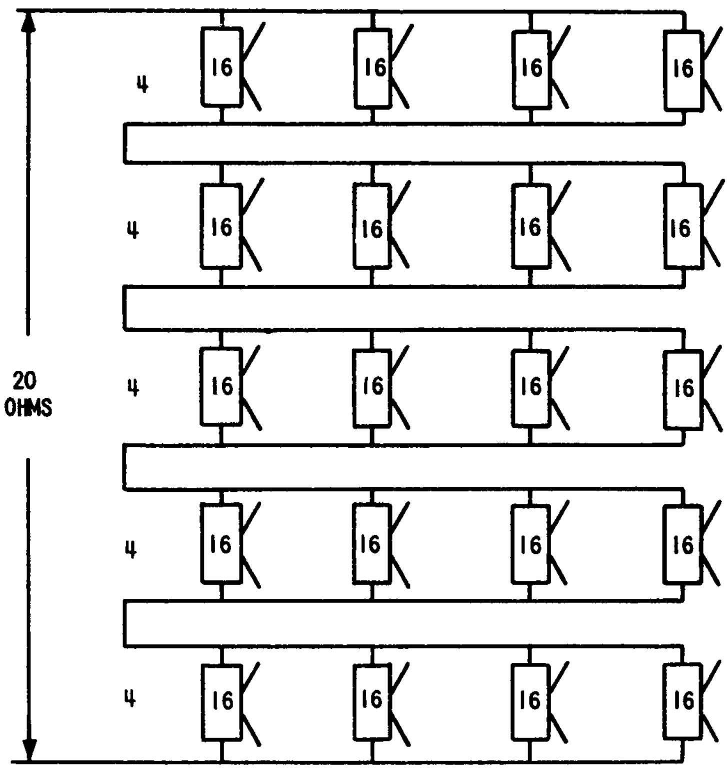


Fig. 3-6. A series-parallel combination of 20 speaker units.

essary, and it didn't make any difference to the amplifier designer what impedance output he provided, why not use an impedance suitable for a constant-voltage distribution system? But now that transistors are here, it is pertinent to re-evaluate the situation.

The output transformer of an amplifier is, far and away, the most costly item on the bill of material. Even in the days of tube amplifiers, some were advocating elimination of the output transformer, although hardly as a cost-saving factor then. The "transformerless" advocates in those days saw the output transformer as a component causing unnecessary distortion. Actually, the distortion an output transformer causes is not serious enough to bother any commercial installation and it's doubtful if any high fidelity addict ever actually heard the difference that was the subject of quite a lot of debate at the time. Eliminating the output transformer from a tube amplifier certainly didn't save cost, or somebody would have done it commercially in those days.

DIRECT AMPLIFIER-SPEAKER CONNECTIONS

The advent of power transistors has changed that. The supply voltages used are not as high as those for tube amplifiers. They can easily be isolated by electrolytic capacitors, or the use of separate supply sections (like voltage doubling, except that the supply provides two separate DC voltages), methods that cost much less than a quality output transformer. So now it's practical, at the amplifier end, to design a distribution system for direct coupling, using impedances in the region of common voice coil impedance. Thus, a re-consideration of the advisability of series-parallel connections is pertinent.

With the improved reliability of modern speakers, a possibility of failure is far less of a detraction. As line distribution transformers are no longer needed, a little extra cost is saved at the speakers. And eliminating transformers makes more complete use of the amplifier's output power. With the system that used an output transformer for the amplifier and a line-matching transformer at each loudspeaker, an appreciable portion of the total power was dissipated in the transformers.

Output transformers run from 90 to 95% efficient; line transformers run from 80 to 85%. When you consider that every bit of audio power had to go through the amplifier's output transformer and one speaker line transformer, some 20 to 30% of all the power got lost on the way. Quite often, that much wouldn't be lost in a direct connection. But most important is the cost saving. Making this change eliminates the most expensive component (and nowadays the most bulky and heavy, as well) in the power amplifier—the output transformer. Although at this writing few amplifiers are yet available for application this way, we predict a progressive swing toward direct connection in future years.

As with most changes, familiarity with current practice prompts a reaction to something new and unfamiliar. One such reaction argues that series-paralleling connections are more complicated. Actually, this is not true. Parallel connections have seemed simpler because that has been the dominant method; the wireman could loop from point to point without giving thought to the kind of connection being made. But in electrical wiring the switch and the appliance it controls have always been in series! Any electrician does that wiring with little enough thought.

Actually, if you weigh the economics, based on the amount of wire involved, there is usually no difference. If all the units are in series, maybe some wire could be saved, sometimes, by using single-conductor in a loop (Fig. 3-7). But a series connection can be used as simply as parallel connection using twin-conductor cable merely by changing the connections at each junction box (Fig. 3-8). For series-parallel, usually a natural arrangement makes it simple to do. Few systems big enough for series-parallel would loop from one end to the other in a straight parallel system. Any large installation usually involves branching connections, if only to distribute the current, and avoid all the current having to be delivered directly to the first speaker on the line.

As long as branching has to be employed in some pattern, it is simple to coordinate it with physical groupings, so each branch serves a section where each unit requires the same level. Then each branch can use parallel connection, with the whole branches connected in series, or vice versa. One

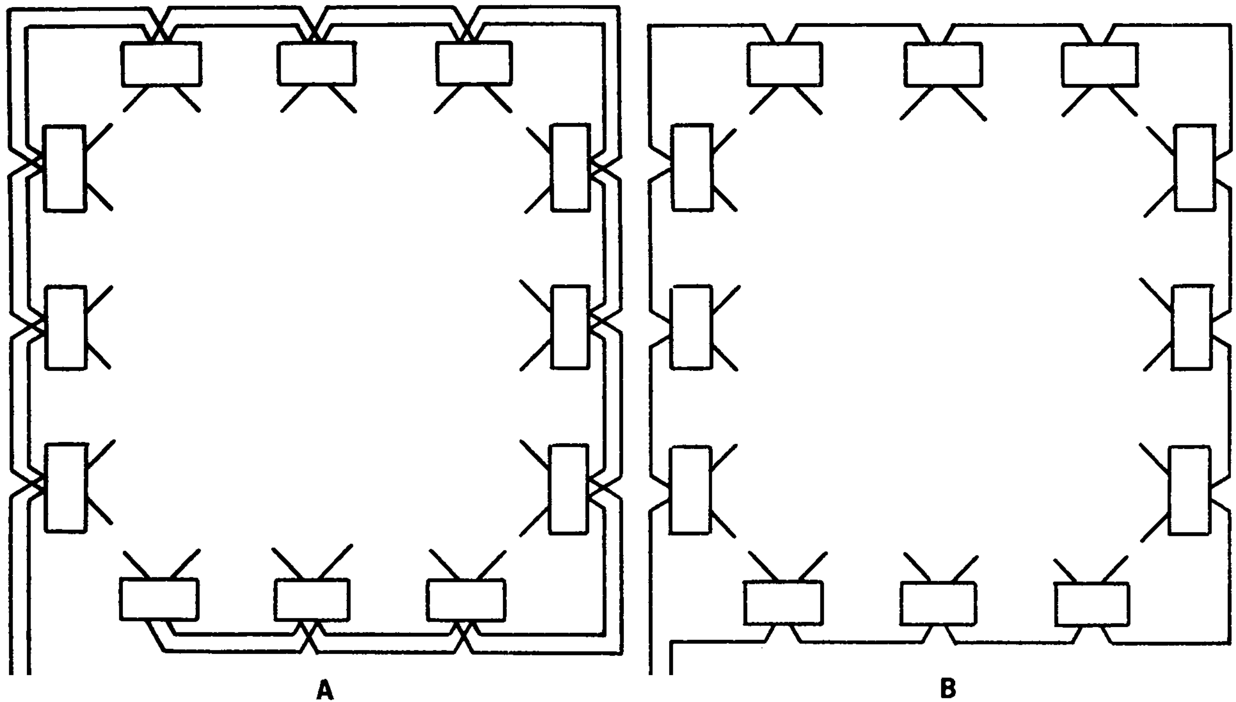


Fig. 3-7. As is apparent in these sketches, by using single-conductor connections, series connection sometimes save wiring (B) over parallel (A).

or the other will invariably provide a workable impedance for that section of the system.

LOW-LEVEL DISTRIBUTION

While this book does not get into the physical aspects of an installation, such as how many speakers and where to put them, the electrical economics may be considered. When a difficult background noise needs overriding by the system, possibly due to a noisy audience, one very necessary thing is to reach all of the audience. Speakers widely spaced, that might serve the entire audience adequately when they are listening quietly, may be quite unsatisfactory except to a few people near each speaker when the audience becomes noisy. The solution to this problem is more speakers, so that every member of the audience is within a shorter distance from the nearest speaker. Then each speaker can be operated at lower individual level, and everyone will hear.

This method, called low-level distribution, has another advantage. One form of background noise is the program emanating from the speakers themselves, after it has turned into reverberation, so as to be no longer intelligible. When this happens, raising the level (assuming you have power available to do so) only aggravates the problem: It makes the program louder, but the noise due to reverberation becomes louder by the same amount—a losing battle.

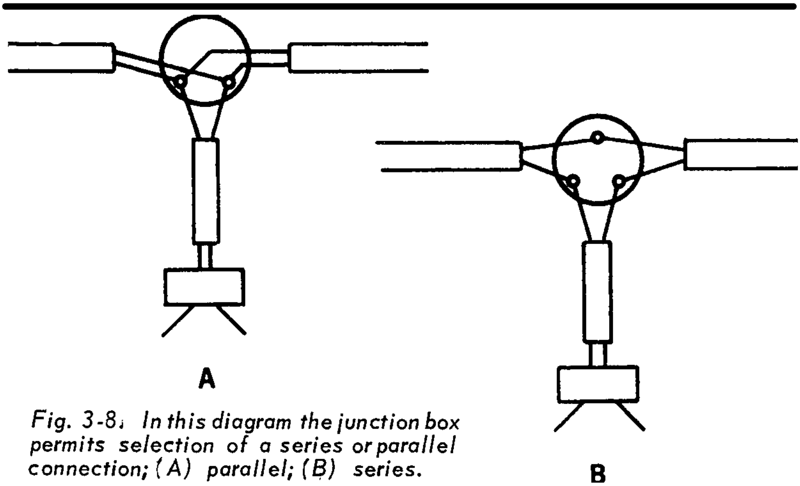


Fig. 3-8. In this diagram the junction box permits selection of a series or parallel connection; (A) parallel; (B) series.

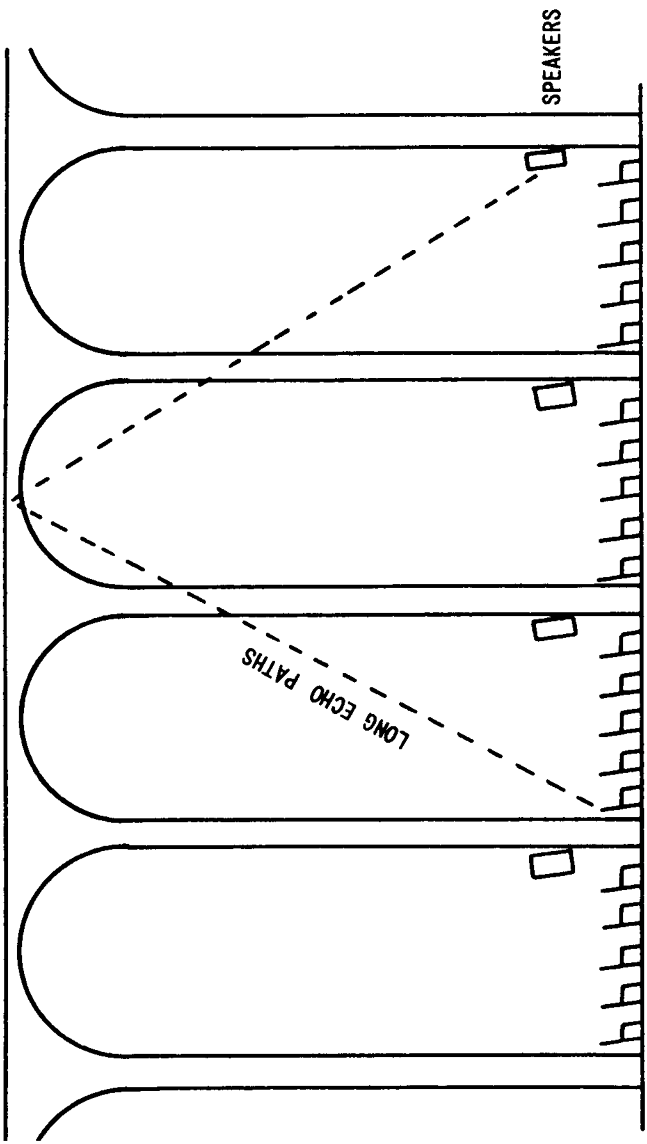


Fig. 3-9. A problem installation: the high roof allows delayed reverberation that causes confusion.

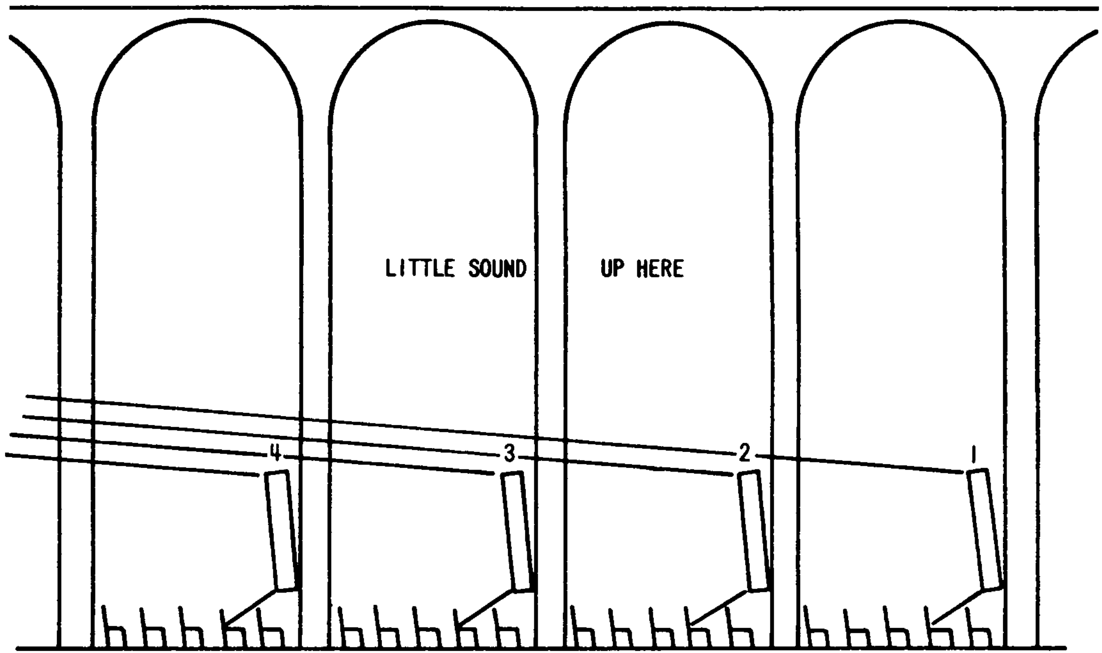


Fig. 3-10. This diagram shows one way to overcome the problem of Fig. 3-9. Column speakers keep most of the sound from reaching the roof. A time delay prevents build-up of other unwanted echo components.

Bringing individual speaker units nearer to the audience reduces the relative effect of reverberation. Each member of the audience is able to hear the speaker nearest him more readily. At the same time, less sound energy is let loose at large to fill the empty spaces that introduce reverberation. So low-level distribution launches a two-fold attack on the reverberation problem.

A particular problem situation is the tall, cathedral-type building. Low-level distribution will help, but the long echo path to the ceiling and back still makes confused sound difficult to overcome. (Fig. 3-9). The use of directional speakers (the column type) avoids sending that much sound up to the roof for a massive echo. But in some instances the acoustics of the building is so bad that even this much care in the selection and placement of speakers is not sufficient, because of reflections from the high walls.

The only solution here is a progressive time delay, so only the direct sound from the person speaking is augmented by the system. In this way the reflected sound from the walls is not augmented, and so reverberation is minimized. To achieve this effect, each rank of speakers must have a successive delay (Fig. 3-10). This means the system must have as many power amplifiers as there are ranks of speakers,

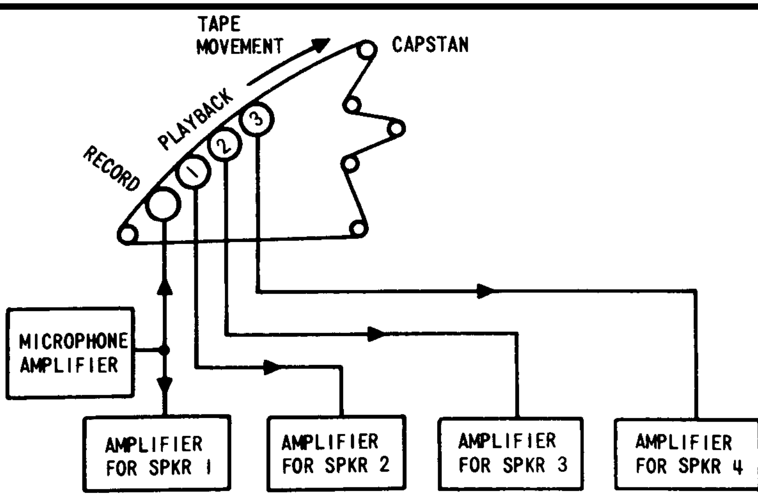


Fig. 3-11. Tape delay unit for use with the system in Fig. 3-10. The speaker group numbers relate to the same numbers on the delay unit.

and then some form of delay is used so the program is fed to each power amplifier input in time sequence. The simplest way to achieve this is with a tape delay loop (Fig. 3-11). A tape, running at fairly high speed, so short time intervals can be represented by successive pickup heads along the tape path, records the program, then plays it back in each playback head in turn. After the last pickup has been made, the tape is erased for re-use the next time around. More sophisticated ways of resolving other distribution problems appear in Chapter 5.

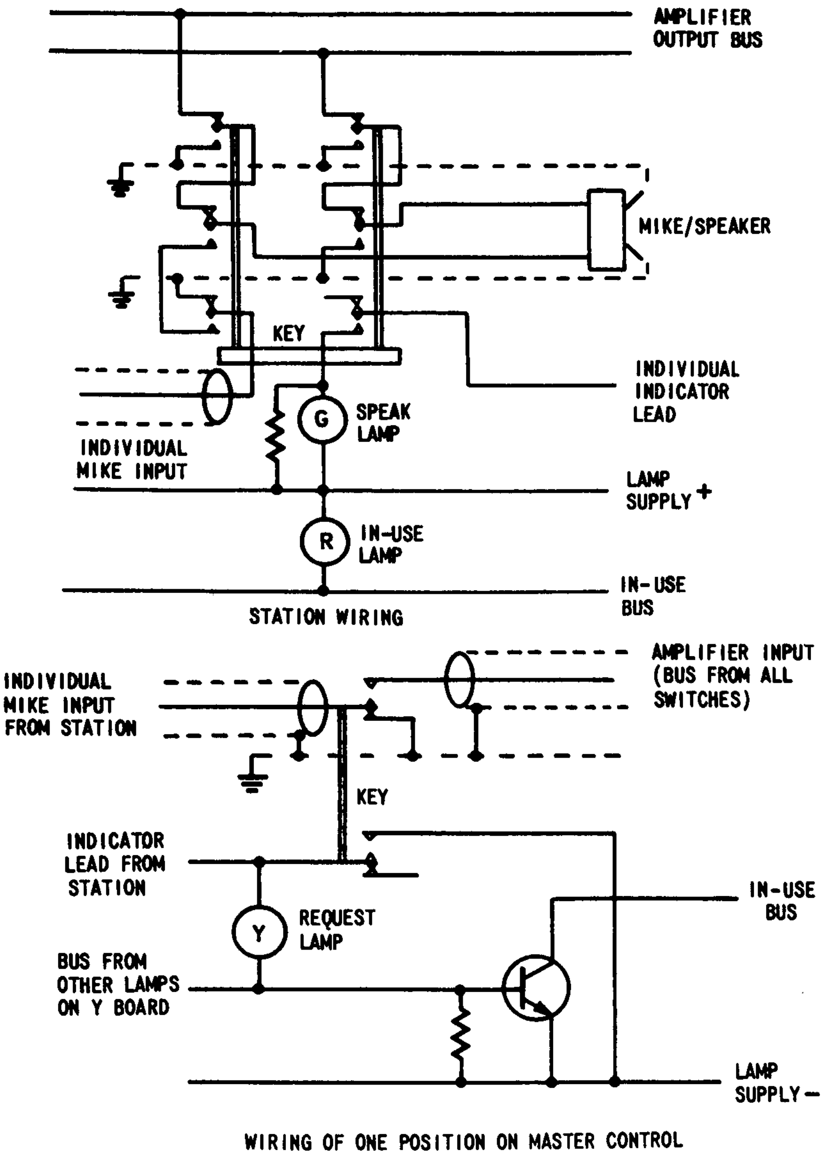
MULTI-WAY SYSTEMS

In some council chamber installations this process has been taken to its logical limit: the provision of an individual speaker for every person present. For that kind of installation, in a room with difficult acoustics (such as some council chambers have) the approach is quite helpful.

As every councilman is allowed the privilege of the floor at different times, his speaker is disconnected when his microphone is in use. The wiring may be arranged so that the small loudspeaker (by switching) also doubles as a microphone. This extends the economy because small loudspeakers cost much less than many high-quality microphones. Such a circuit needs very careful attention to the switching at various locations, because both input and output must be switchable to the same unit, and this can cause unwanted electrical feedback from output to input. Proper use of shielding can care for this.

The speakers used should have a high voice-coil impedance. A matching transformer is not desirable, because when functioning as a microphone its core would be susceptible to hum pickup and the cost of shielding (magnetic shielding) such a transformer would be prohibitive. The speakers should all be parallel connected (when working as speakers) so the wiring to all switches can be balanced to ground.

Another addition to such a system might be an extra contact, with light-gauge wiring, which will operate a light on the speaker's or chairman's desk, to indicate who wishes to speak, thus aiding in recognizing any councilman who may have something to say. A further addition to the circuit can



F . 3-12. Wiring for the multi-way system described in the text. At each station, a red lamp glows when someone is using a microphone. Pushing the button requests use of the microphone. The green light indicates when the master control has given permission by pressing his corresponding button. The requests are indicated at the master control by yellow lamps, one for each location.

provide that each position receives an indication that someone else has pressed his speak key at the same time, when this happens, as well as an indication to the one selected by a button at the speaker's desk to be allowed to speak next (Fig. 3-12).

Chapter 4

Program Sources

An important fact to remember about any program source, be it microphone, radio, phonograph, tape, etc., is that it provides the first limitation to the whole system: The system can handle only what comes to it from this source. Immediately, this justifies spending more than what might seem like a proportionate amount on such items, but it also may lead to the false assumption that cost is a direct index of quality, or of suitability for the job at hand.

MICROPHONES

The layman—and many a technician too—sees the microphone as a relatively simple device that may be good, bad, or indifferent. A more sophisticated evaluation may go a little further and say that microphones may vary in two principal properties—sensitivity and quality. But, even that is an over-simplification. Chapter 1 shows that sensitivity depends on matching. For the microphone to perform according to the sensitivity stated in its specification, its impedance must work effectively with the input impedance of the amplifier to which it is connected.

SENSITIVITY

That term "sensitivity" is sometimes misleading. By analogy with hearing, one is apt to think of sensitivity as the inherent ability to pick up sound; that a more sensitive microphone would be able to pick up softer sounds, or those coming from a greater distance. In the basic sense, this is true, but other factors also contribute to a determination of the operative sensitivity. The word "sensitivity" as applied in the technical specification of a microphone merely tells how much of the acoustic energy arriving at its location is converted into electrical impulses that can be amplified. The level to which they are eventually amplified, which also affects the apparent pickup range of the microphone, depends on the gain of the amplifier.

Two scales of sensitivity ratings are used for microphones. For low-and-line-impedance microphones the reference is dbm for either 1 millibar (dyne per square centimeter) or for 10 millibars. The latter is an average sound intensity from someone speaking 8 to 10 inches from a microphone. You may need to take a lower figure for soft-voiced individuals. For one or the other of these standard sound pressures, the output of a low- or line-impedance microphone is rated in dbm. For high-impedance microphones a minimum loading impedance is usually specified, e.g., 50,000 ohms, and the output level at one of the standard sound pressures is rated in db referred to 1 volt, because power reference is not meaningful where the impedance may be vague.

Correspondingly, amplifier inputs are rated at the level needed to produce full output, using dbm for low- and line-impedance inputs and db referred to 1 volt, or an actual figure in mv, for high-impedance inputs.

Example. A microphone is rated at -54 dbm with 10 millibars, 250 ohms. The same microphone might be rated at -74 dbm with 1 millibar. -54 dbm is 2 millimicrowatts. In 250 ohms this is 0.7 mv. An amplifier rated to give full output, with 250-ohm input, at either -54 dbm, or 0.7 mv, or less, will work with this microphone to give full output from someone speaking at 8 to 10 inches.

Example. A microphone is rated -54 db referred to 1 volt, high impedance, 10 millibar input. This means an input of 2 mv. An amplifier with a correspondingly rated high-impedance input would give full output from someone talking 8 to 10 inches from a microphone.

The above is not all relating to the capability of a microphone and amplifier to pick up sound. It may summarize the needs, but it omits some obstacles. The amplifier may have plenty of gain, but an "acoustic howl" starts before you can use very much of the available gain, so that the usable sensitivity of the microphone is still not as good as you might want.

The tendency to produce acoustic feedback, or howl, depends primarily on building acoustics, but microphone qualities and other parts of the audio system can have a pronounced effect on it, too. Some microphones seem to start an acoustic howl before they will pick up someone speaking relatively close to them, while, in the same situation, another microphone seems to reach out and pick up sounds, without producing a howl.

What is the difference? There is no simple answer. Two major properties of a microphone contribute to this kind of difference: frequency response and directivity.

FREQUENCY RESPONSE

Howl occurs at only one frequency, or predominantly at one frequency—the first one to "get going" forms a standing-wave pattern and it "goes" from there. A microphone with a "peaky" frequency response is more likely to "find" such a frequency before it picks up the many frequencies contained in a person's voice (or whatever program is being picked up) than a microphone that picks up all frequencies relatively uniformly.

In any frequency response specification there are two important factors: (1) frequency range and (2) deviation in sensitivity throughout that range. We may illustrate this by comparing the response of two hypothetical microphones (Fig. 4-1). Microphone A is specified as "flat" within ± 2 db from 150 to 9,000 Hz. Microphone B is specified as flat within ± 6 db from 50 to 15,000 Hz. Which represents the better microphone? Only the complete curves can tell. Sometimes these two specifications could relate to the same microphone: its response might stay within ± 2 db from 150 to 9,000 Hz, and deviate as far as ± 6 db in the extensions from 150 Hz down to 50 Hz and from 9,000 Hz up to 15,000 Hz. The only sure way to tell the difference is to compare curves, taken

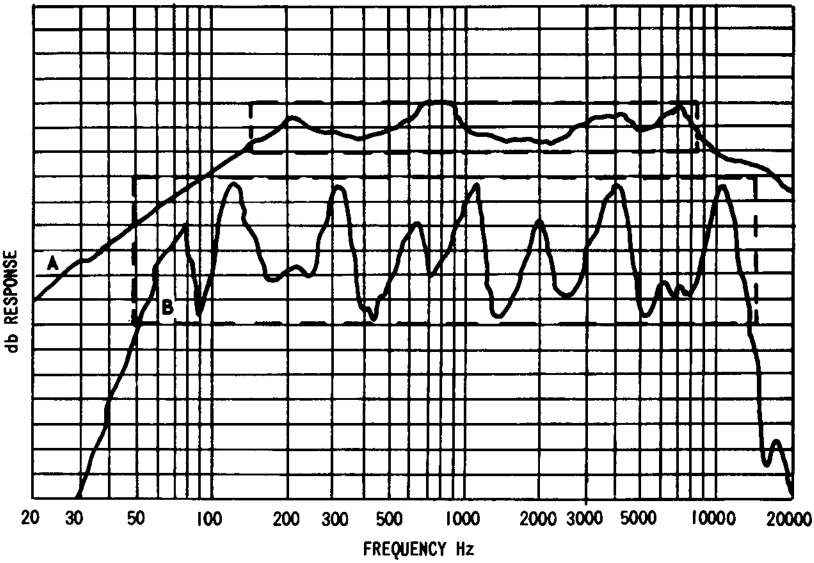


Fig. 4-1. This frequency response comparison between two microphones illustrates significant variations in total response at certain frequencies.

by a reliable authority. But assuming the specifications represent typical data, accurately taken, we can be reasonably sure that the manufacturer will use a specification that presents the best aspects of his microphone's performance.

Response A in Fig. 4-1 obviously represents a microphone which has been designed to achieve minimum deviation—within ±2 db. To avoid acoustic howl, this is important. How much sound intelligence will you want to pick up below 150 Hz, or above 9,000 Hz? Not much, in all probability. But if you do, how does the microphone perform out there? The response may extend some distance on the frequency scale before exhibiting even ±6 db deviation, or it may widely exceed those limits not far outside the specified frequency range of 150 to 9,000—usually cutting off (Fig. 4-2). The simple specification doesn't reveal this.

On the other hand, Response B obviously represents a microphone where the designer aimed at achieving maximum frequency range. The frequency figures, 50 to 15,000, look good. The ±6 db deviation does not look so good. If you took

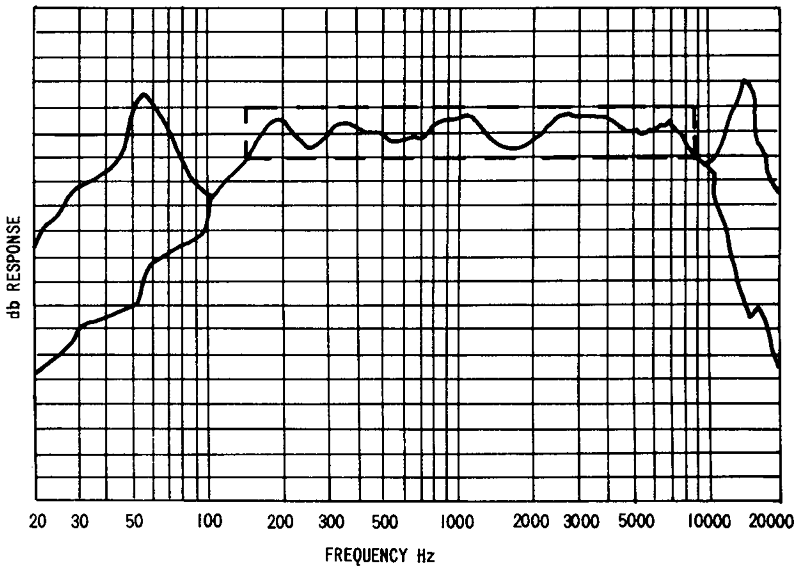


Fig. 4-2 From Fig. 4-1, Response A could be represented by these typical variations.

a smaller range, say from 500 to 5,000 Hz, which is not a good range at all, the same microphone might still show a deviation of ±6 db.

From this discussion it is evident that, for most purposes, the flatness of the response, meaning how little it deviates, is more important than the width of the frequency range it covers within that deviation. Any sound system man will avoid buying a microphone where the manufacturer merely stays, "response from 80 to 12,000 Hz," without specifying what response. Such specification data is meaningless.

DIRECTIVITY

Directivity sounds like a simple quality, but this too is more complicated than it sounds. It helps prevent howl by emphasizing the wanted sound more than the sound returning from the speaker system, and it also improves the pickup of wanted sound as compared with unwanted background noise. But the relationship by which directivity makes these changes is by no means a simple one.

The original form of directional microphone was the ribbon, which is bidirectional; it has maximum pickup in two directions, and a plane of null, or silence (Fig. 4-3). By using one (or both) of the major pickup directions and arranging that as much as possible of the unwanted pickup or source of howlback is in the plane of silence, better discrimination for the wanted program against unwanted interference is achieved. If unwanted pickup is uniformly distributed in the directions from which it arrives at the microphone, the use of a bidirectional microphone will improve discrimination by about 8 db. If the unwanted sound is more strongly concen-

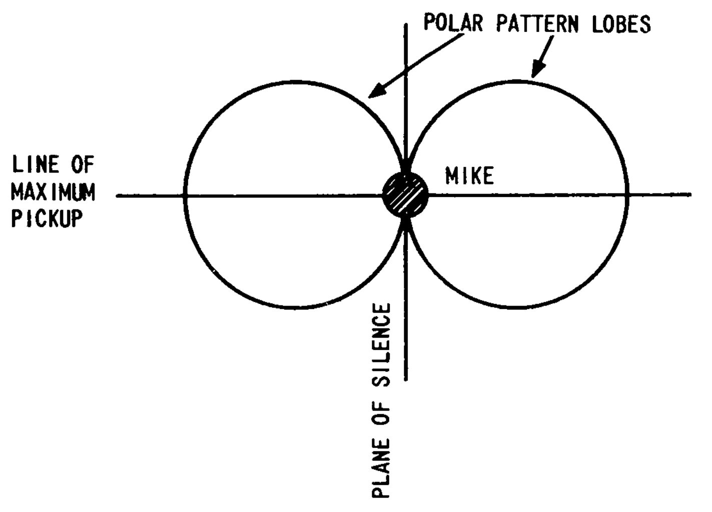


Fig. 4-3. Pickup pattern of a bidirectional (ribbon) microphone.

trated in the rejection plane, the improvement can be greater than that.

The second form of directional pattern is the cardioid, or unidirectional. This eliminates one of the directions of maximum pickup in the bidirectional, and doubles the other. It has a null on the side opposite to the direction of maximum pickup (Fig. 4-4). For random unwanted sound (distributed uniformly from all directions) the cardioid is not better, and sometimes not as good, as the bidirectional type. However, where one specific direction gives trouble, such as behind the microphone, it can be much better. For example, on stage, the cardioid can effectively reduce feedback from the

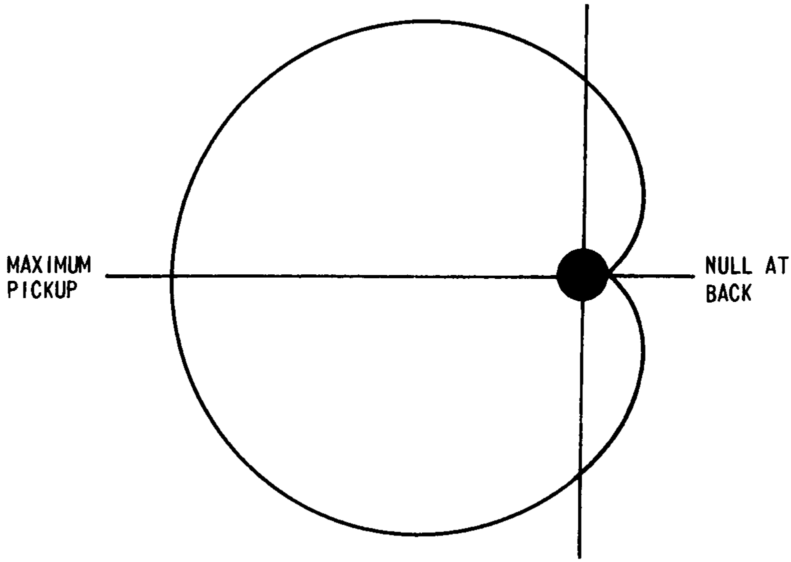


Fig. 4-4. Pickup patterns of a cardioid (unidirectional) microphone.

auditorium (Fig. 4-5). Each type of directivity has its applications for which it is particularly well adapted.

For many applications a pattern that falls between the two, called a "super-cardioid" (Fig. 4-6), does the best job. By allowing a small pickup range directly in back, the width of

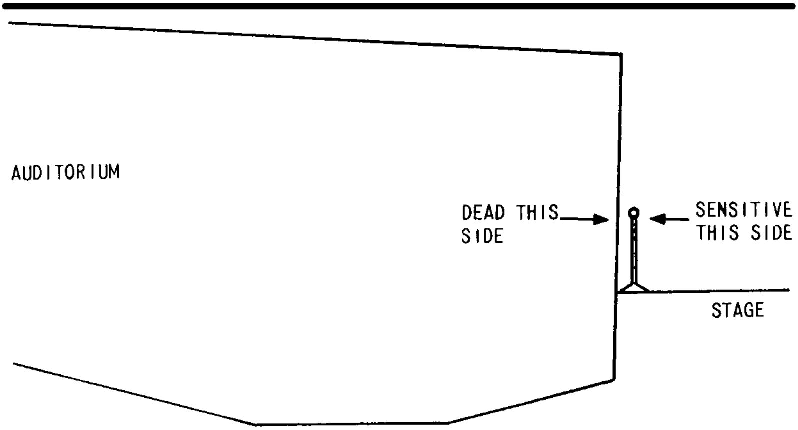


Fig. 4-5. Here is a situation where cardioid response serves better than the bidirectional.

the front pickup zone is reduced and the sensitivity in the back also is reduced considerably. Also derived from these basic forms are more sophisticated microphones, rather like the super-directional antennas used for radio reception. The super-directional microphone can pick out sound from one particular direction, against quite high general background noise.

Like frequency response, directivity is not always a simple property. In simple description, we designate a zone or area as "dead," without sensitivity to pickup, while another

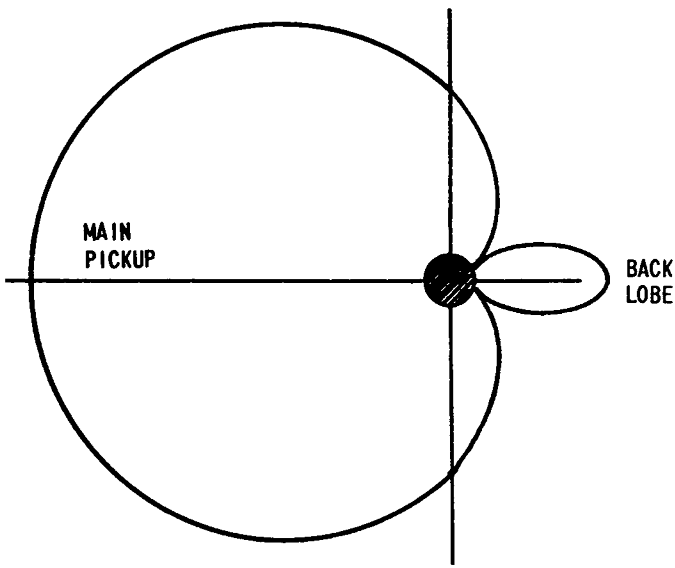


Fig. 4-6. A typical super-cordioid directional pattern.

area produces maximum pickup. But the performance may not be that simple. If all sound was of one frequency, such clear-cut separations would be quite possible. But sound covers a spectrum of between two and three decades of frequency. Few so-called directional microphones retain the full degree of discrimination down to the lowest audio frequencies. Fortunately, directivity at the extreme low end is not often vital and, when these frequencies cause trouble, a bass cut will usually take care of it.

Also, few microphones retain the precise directional char-

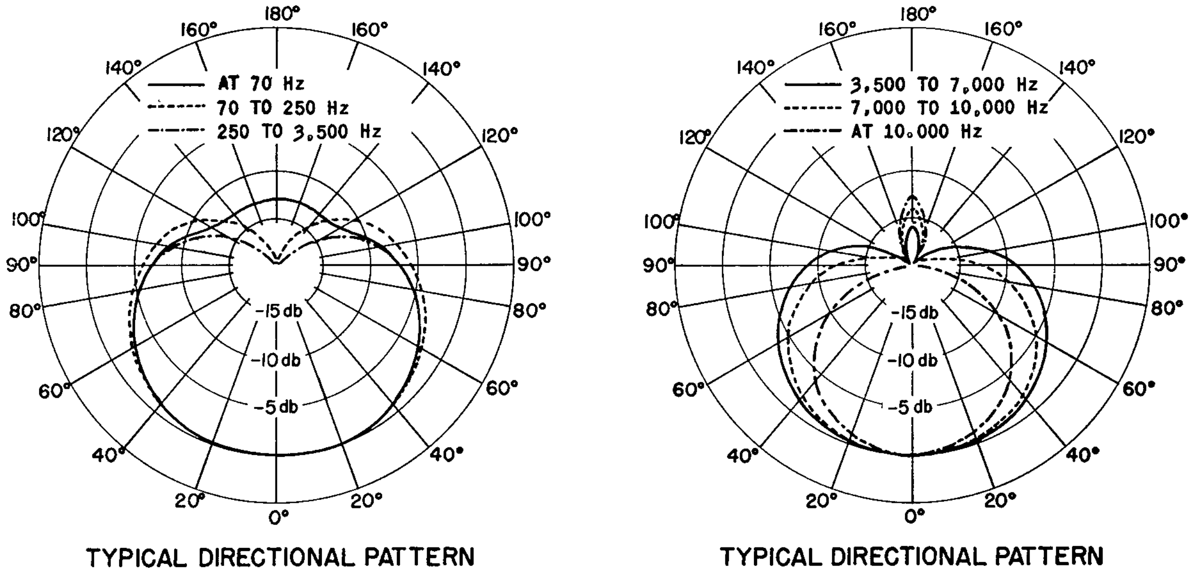


Fig. 4-7. Actual polar patterns of a microphone, taken at different frequencies (Courtesy: Shure Brothers, Inc.).

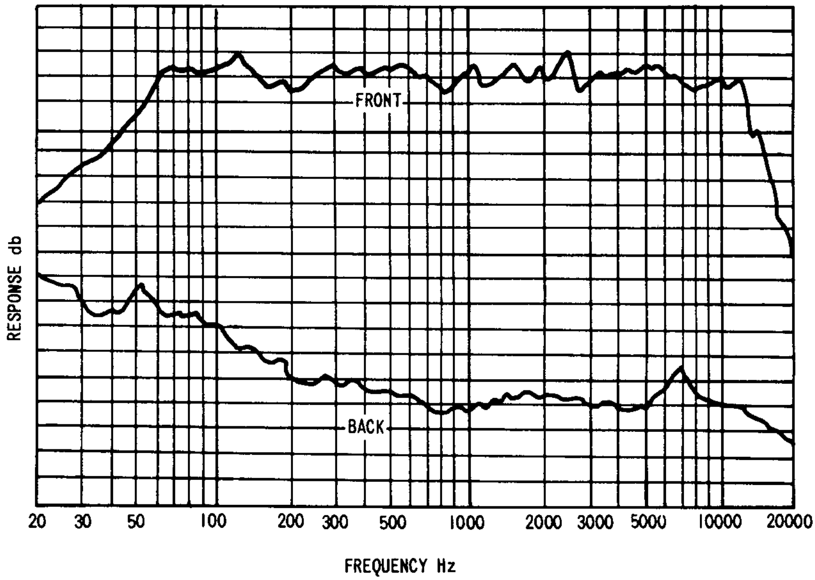


Fig. 4-8. An alternative way of presenting directivity data. These response curves represent a good unidirectional microphone.

acteristic at the high-frequency end. But through the middle-frequency range, where most of the problems happen, a good directional microphone should maintain a good polar response (Fig. 4-7). Or expressed another way, the discrimination in different directions should be reasonably maintained throughout most of the frequency range (Fig. 4-8). A poor quality microphone may not do this (Fig. 4-9). In this case, the microphone's directional properties are of little value. The frequencies at which they are good do not help overall performance, because howlback just finds other frequencies where the mike isn't so good. A good frequency response, measured in the "wanted" direction, is no guarantee of good directivity. But a microphone with a poor frequency response will almost inevitably have a poor directivity response.

A non-directional (also called omni-directional) microphone is completely non-directional only over the lower frequency range. At higher frequencies it loses response from behind by sheer obstacle effect—getting in its own way, so to speak. Thus, a high quality nondirectional microphone may sometimes perform better, at least in some circumstances, than

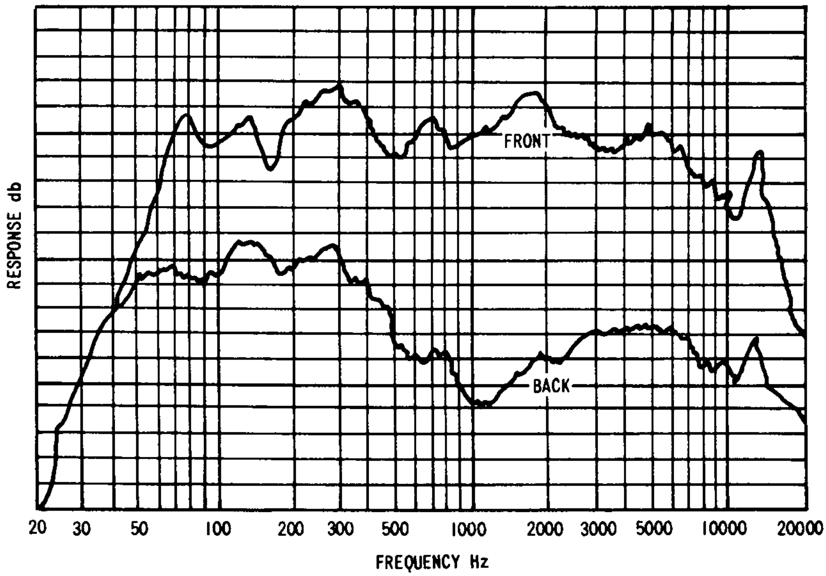


Fig. 4-9. Response curves of a microphone considerably inferior to that represented in Fig. 4-8.

a poorer quality directional type. Remember that both a smooth frequency response and the directional properties fight the problems. So a better frequency response may do more than improved directivity in the mid-range.

AUXILIARY AMPLIFIERS

Another part of the problem in good microphone utilization is achieving adequate control over their pickup. First, microphones used for long-range pickup need more amplification than that provided for relatively close talking. The long-range pickup often calls for auxiliary preamplifiers. For pickup over a large area, such as the entire stage used for a play or some other production, a single microphone is not close enough to any of the performers for good intelligibility. The sensitivity may be adequate, but sound reflections become as loud, or almost so, as the direct sound. Consequently, the reproduced sound appears very echoic.

In assessing this during a survey of a situation, don't forget that a microphone is a "one-eared" device. Unless you, your-self, have a defective ear, your binaural hearing faculty is subconsciously able to help you in a manner that a microphone cannot achieve for its listeners. In a poor acoustic environment your binaural faculty can discriminate between direct and reverberant sound in a manner the microphone cannot. Once the picked up sound is in the system, you no longer can discriminate between sounds from a wanted direction and all the others in the reproduced sound. This fact accounts for what seems an exaggeratedly echoic effect in some reproduced sound. It is also why recording engineers prefer studios that are on the "dead" side and add reverberation "to taste" artifically afterward, rather than accepting the natural reverberation of a "live" studio: you can put it in, but you can't take it out!

"Close miking" overcomes this problem of exaggerated echoic effects in situations where deadening the studio or stage is not practicable. But using many microphones, all live at once, can exaggerate the echoic condition electrically, rather than acoustically, by using multiple pickup of the same sounds, with time differences, depending on the distances of individual mikes from the sources of sound. So to benefit from the use of multiple mikes, it is desirable that only wanted sound should be picked up at all times.

Where multiple miking is used to cover a wide area of pickup, all of them should be "gain-ridden" for optimum effect at all times. When all the instruments of a full orchestra are playing together, that is the only time when all microphones should be operating at approximately the same gain (which is less than any of them could operate individually, because acoustic feedback tends to be cumulative when multiple mikes are in use). At all other times, such as when a solo instrument or group of instruments is being featured, the appropriate microphone(s) should be brought up a little, and the rest played down, or turned off. Skilled operation by someone, preferably with a good cue sheet, but imperatively in a position where he can watch the program closely, can produce an infinitely superior result.

The same is true of a dramatic performance, perhaps even more evidently. Watching a play involving a roomful of characters scattered across the stage, you may get the impression that they all "talk at once." Actually, they don't, or only very rarely do they do so. The sound impact is planned much like an orchestral production. The center of attention invariably turns to one player at a time, who should have preference in the mike level adjustment at that moment. If some secondary conversation is going on as background, the level of the mike to pick that up should maintain audibility, but not equally so with the principal parts. This is a good operator's job. And generally this kind of operation will require an auxiliary control where the operator can see what's going on.

REMOTE CONTROLS

In some instances, a remote may be useful. This is not the same kind of remote discussed in the next Chapter. In this remote the actual controls are effected electronically in an amplifier that can be stowed anywhere, preferably to suit the convenience of microphone wiring and main system inputs. But the operation of these controls is effected remotely by voltages from controls at a location where the operator can see what's going on, and/or be in close touch with the director. The principle on which a remote mixer works is illustrated in Fig. 4-10, where a variable voltage is applied to some circuit that responds to it by changing the gain. In a tube circuit, a variable-mu pentode with a short grid-base (such as a 6J7) is used to change gain. This is not so easy for transistor circuits, because transistors do not exhibit this kind of curvature.

The best characteristic to use with transistors is their variable AC resistance. To do this, select a transistor where the slope of the collector characteristics opens out, rather than being constant (Fig. 4-11). Then DC controls are applied so the voltage across these transistors is maintained substantially constant, while the current, and thus the AC resistance, is varied (Fig. 4-12). To do this requires a parallel connection for voltage, with series operation for signal. The current divides between the two transistors so the total remains substantially constant. It is useful to employ a third transistor as a constant-current "load" to feed the DC to both the control transistors.

This constant-current transistor should have as high an AC resistance as possible to avoid reducing the effectiveness of the control transistor's AC resistance variation. When proper operating conditions are selected a transistorized gain control will work as effectively as the tube type, but it takes several transistors where only one tube is required. This kind of remote can be made far more flexible than some direct controls. Different performers may require different tone compensation for best intelligibility or most convincing or natural effect. This can be provided by remote tone control, operated by switching that alters microphone settings at the same time.

Such a remote control can be provided with duplicate, or even triplicate controls and a selector switch that changes

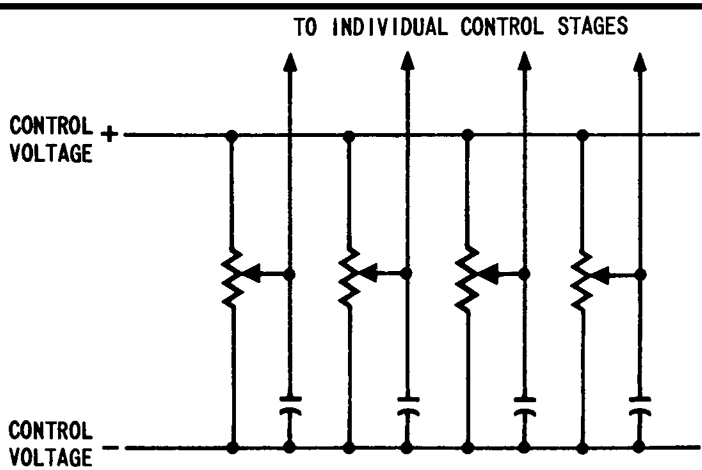


Fig. 4-10. A simple remote control circuit, using electronic gain variation in equipment more suitably placed for the system.

from one to another, as required. In this way each group of controls can be set for a particular part or player's performance and they can be changed back and forth with minimum effort (Fig. 4-13). It takes extreme skill and concentration to alter several controls at the same time, and even more to do so repetitively, as during a conversation. It is much simpler to set up such changes so a simple throw of a key switch effects all necessary changes at the same time.

A more sophisticated system, developed by Fairchild, uses light as a means of control. Constant-impedance attenuators, whose value is controlled by variable light, replace the usual type where some form of sliding contact is used. This means that the circuits controlling the supply for the variable light

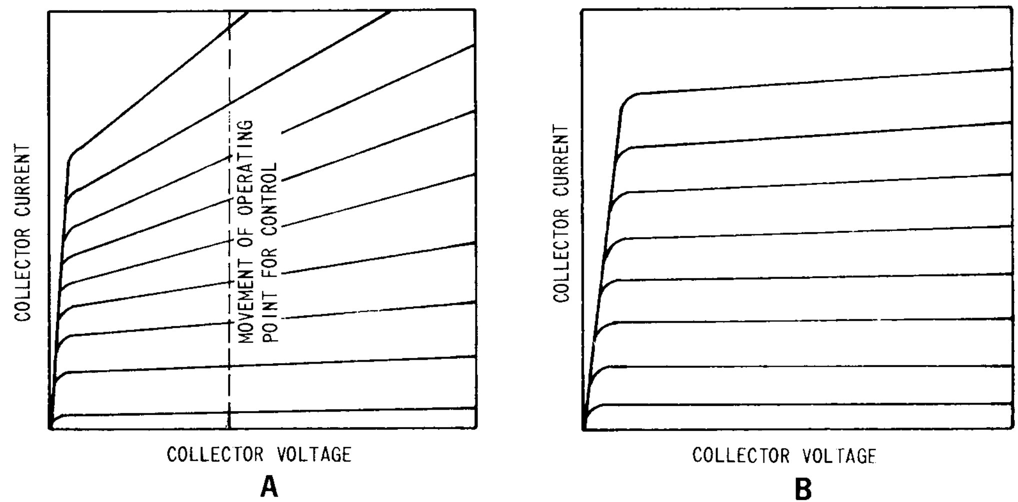


Fig. 4-11. This comparison of transistor characteristics shows the type suited for a variable-gain stage: They should open as at (A), rather then being parallel, as at (B).

can be remoted. Also the change in attenuation occurs quite smoothly and noiselessly.

RADIO FEEDS

Many programs originate from radio hookups these days. The basic source here, at the local level, is a reliable radio receiver. Less reliable than the receiver may be the reception conditions. So, if at all possible, it is good to plan for alternative radio routing. This is especially true where transmission paths cross mountains.

Weather conditions may completely interrupt one transmission path at times. If another one is open, or an alternative point of origin (itself perhaps another pickup point in the network or hook-up) can be used, the program may continue where relying on only one transmission path could result in a loss of program. Communications-type radio receivers can do much to minimize interference. This is outside the scope of the present book, because it is not audio, but such facilities should be used to ensure a good source of reliable audio, or as good and reliable as possible, anyway.

PHONOGRAPH AND TAPE PICKUP

Phonograph and tape program sources may be used either separately or in combination with other sources. For example, a dramatic production may use live players, but require background music or other sound effects that could be "canned," using either phonograph or tape as a source for the effects.

Where there is little time for advance preparation, a phonograph is probably the easiest to use, provided all the needed forms of recording are available. It is relatively easy to drop the stylus on a spot in the record carefully located beforehand. In any event, it is easier than finding a particular spot on a prerecorded tape that has not been made specially for the purpose.

Where there is time to prerecord a tape specially for the particular program or presentation, it is well worth the effort involved. Each sound effect required can be recorded on the tape, in the order needed, and the special tape can be simply

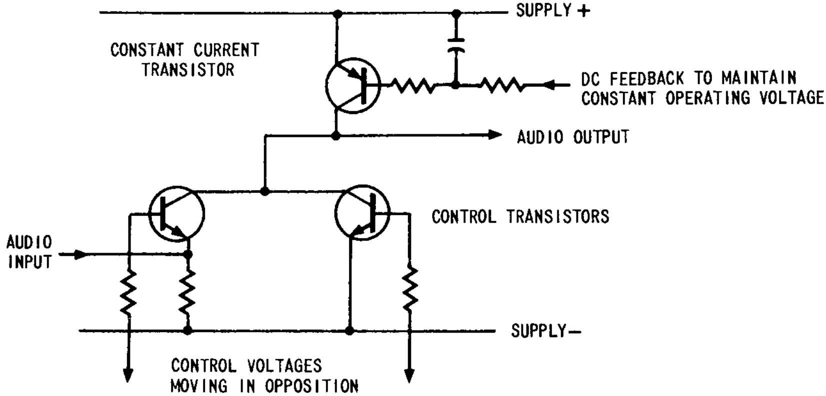


Fig. 4-12. The basic elements of a transistorized gain control.

started and stopped as each successive effect is needed. But this requires careful planning, positive finalization of the program (no ad-libs allowed, or last-minute changes), and detailed advance preparation of the tape. Tying a disc or tape recorder into the system is relatively simple, and involves only the mixing discussed in Chapter 3, with particular attention to seeing that all parts of the composite program are brought together at the correct levels, and all correctly equalized, according to the needs of each part, and with properly matching impedances.

Phonograph pickups come in two major varieties, designated

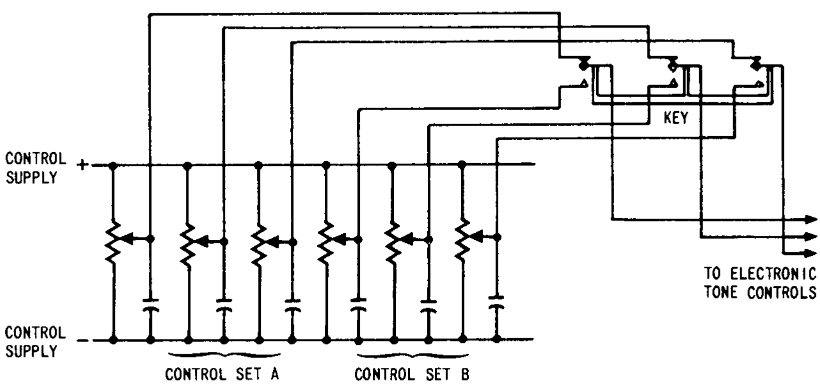


Fig. 4-13. A remote control, such as that diagrammed here, provides quickly switchable alternate controls for gain or tone.

according to their functional difference, or according to the type of transducer used. Functionally, one group produces an output proportional to the velocity of lateral movement. Pickups that function this way use some form of electromagnetic transducer, either moving coil, variable reluctance, or moving magnet. The other variety of pickup produces an output proportional to the amplitude of lateral movement. These use piezo-electric transducers. Strictly, the open-circuit voltage from such a pickup is proportional to instantaneous lateral position. But when this pickup is fed into a resistance of such value that its source capacitance controls the output current (rather than voltage) it behaves as a velocity pickup.

Most professional pickups belong to one of the velocity types and the moving magnet type seems to have proved itself the best design, in terms of achieving the lightest necessary tracking force of the stylus on the record, and in terms of producing the most perfect frequency response, freedom from distortion, and (in the case of stereo pickups) separation.

Pickup sensitivity is rated in output level for a specified velocity of stylus movement. The figure given is the maximum velocity during a sinusoidal movement at specified frequency, and can be calculated from the amplitude and frequency. However, as velocity corresponds to the angle at which the cut moves laterally, regardless of frequency, a constant velocity cut produces a constant width band of light when viewed by reflection from a point source (Fig. 4-14). In fact, the recorded frequency response of a test disc on which a gliding tone has been recorded can be determined by visual examination. This also will show distortion (Fig. 4-15), because the band will lack symmetry and possibly show an irregular thickening of the light pattern.

The equalization specified in Chapter 2 for disc recording is based on the use of a velocity pickup, so any of these pickup types require equalization of that form to yield an output that is a replica of the original input. As most of the velocity types have an impedance that includes an inductive component, one way of getting high-frequency roll-off (at 2,120 Hz in the American equalization) is to load the pickup with a resistance value so that its internal inductance provides the required roll-off. Then the amplifier circuitry merely has to provide the low-frequency boost (starting at 500 Hz on the American characteristic).

While ceramic type pickups, which are basically amplitude type, are not often used professionally, their quality is much better than earlier specimens of the type. Further, they do give a higher output and are less costly than the electro-

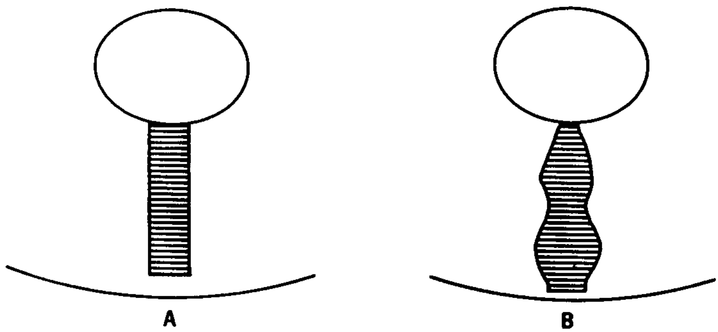


Fig. 4-14. As shown here, reflected light patterns may be used in determining recorded characteristics on a phonograph disc: (a) constant velocity; (b) a response curve that deviates from constant velocity.

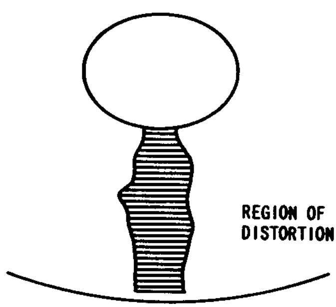


Fig. 4-15. Here is how distortion shows on a light pattern reflection.

magnetic types, at least for corresponding quality. Thus they have an appeal for application in a system where cost is a factor. And saving here does not have such a serious effect on overall performance as buying a "cheap" microphone would have.

To equalize one of these the pickup is loaded with a combination of resistance and capacitance, based on the correction required for a constant amplitude output, rather than a constant velocity output (Fig. 4-16). For example (Fig. 4-17),

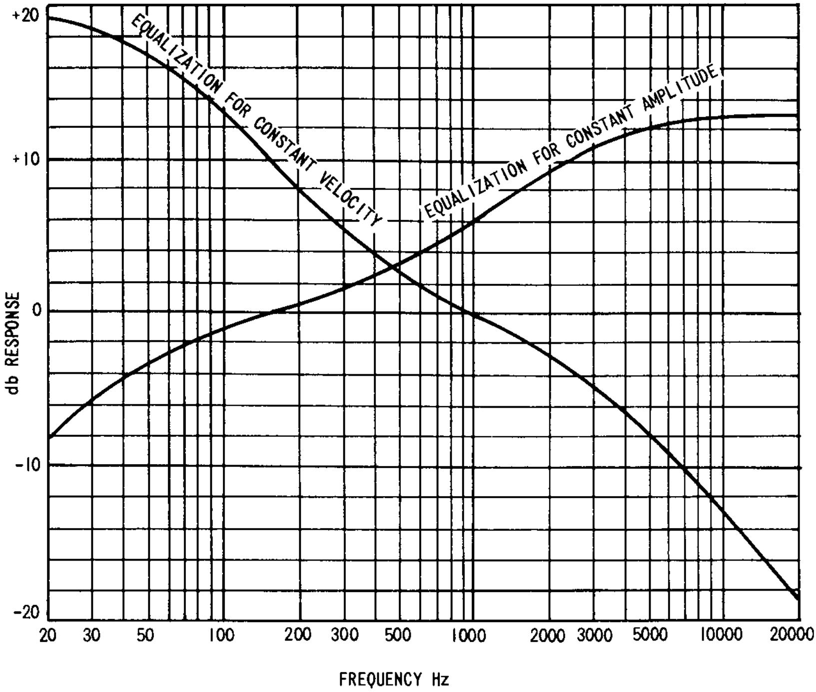


Fig. 4-16. Equalization curve for constant-velocity playback (repeat of Fig. 2-3, American curve) and for constant - amplitude, formed by changing the slope by 6 db/octave.

if the internal capacitance of the piezo element is 0.001 mfd, a total resistance of about 3.2 megohms will produce a turnover at 50 Hz. Now we require a boost that starts at 500 Hz and finishes at 2,120 Hz, for which the 3.2 meg needs to divide into 750K and 2.4 meg. Bypassing the 2.4 meg with a 133 mfd capacitor will give the required boost.

All tape recorder heads employ the electromagnetic principle, for the very simple reason that the device itself is of this character. With both phonograph (when electromagnetic heads are used to record and playback) and tape recording, the following relationship exists, and needs to be understood to establish correct functioning both ways. The recording is referenced against output current from the drive amplifier. The cut made by the stylus in disc recording and the intensity of magnetization impressed on the tape are each referred to the audio current fed to the record head.

Playback is referenced to induced voltage. Pickup output voltage is referenced to velocity of stylus movement. Playback head output voltage is referenced to the rate of magnetic flux change density on the tape. Thus, in both systems the overall effect, viewing the recorded medium as merely a method of storing indefinitely "what happens" (program), and comparing "what goes in" when the recording is made with "what comes out" when it is played back, the input takes the form of current that produces a certain voltage output (Fig. 4-18). This relationship tends to have the properties of an inductance.

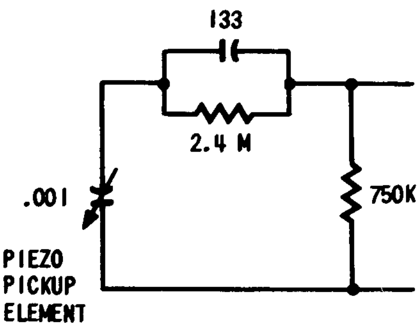


Fig. 4-17. Equalization circuit for a ceramic pickup with an internal capacitance of 0.0001 mfd.

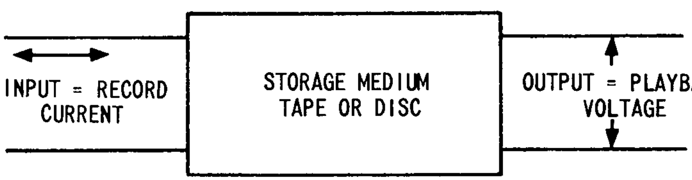


Fig. 4-18. The basic electrical "dimensions" of a recording medium.

ELECTRONICALLY-GENERATED SOURCES

Modern audio systems are called on to provide some extras that weren't even considered to be features of an audio system a few years ago. The nearest thing would be sound effects, which usually were provided by special phonograph discs and careful cueing. But nowadays many new sounds are generated electronically.

The basic element in most of these devices is the multivibrator (Fig. 4-19). This generates a square waveform at the collectors (since the transistor version is so much more adaptable, we'll not spend time on the tube version). Its frequency can be controlled by applying a variable voltage to the top end of the base resistors (Fig. 4-20). When the voltage at the top end of the base resistors is the same as that at the top end of the collector resistors (as in Fig. 4-19) the half period of the square wave is almost exactly 0.7 of the time constant resulting from the base resistor combined with the coupling capacitor. Thus, if R2 is 10K, C is 1 mfd, the half-period is 7 milliseconds, or the full period 14 milliseconds, corresponding to a frequency of about 71 Hz.

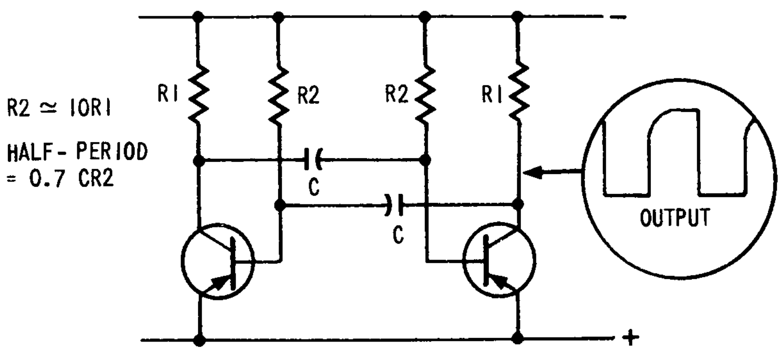


Fig. 4-19. Basic multivibrator tone generator circuit for general use.

Changing R2 or C will change frequency. And frequency can be made variable by changing the voltage at the top of the base resistors without changing that at the top of the collector resistors. Raising the supply voltage to the base resistors raises frequency; lowering it, lowers frequency, until a point is reached where the transistors do not maintain saturation for the entire half-period, when waveform changes and frequency begins to rise again.

Amplitude can be controlled by passing the generated square wave through a partially saturated stage (Fig. 4-21) and changing the degree of saturation. Complete saturation blots out the signal. Reducing the saturation increases signal, until reducing the current from the source providing saturation (lowering the control voltage) to zero allows a full amplitude signal to pass.

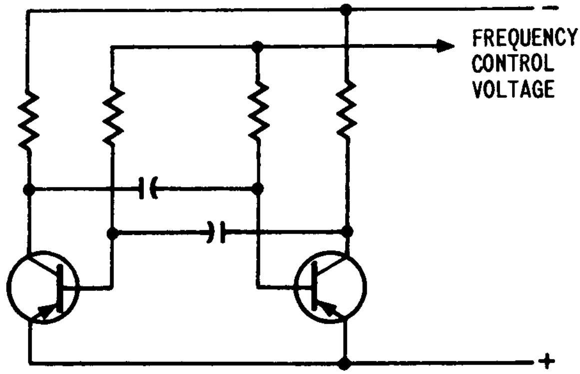


Fig. 4-20. With the proper component changes, the multivibrator frequency can be made to vary.

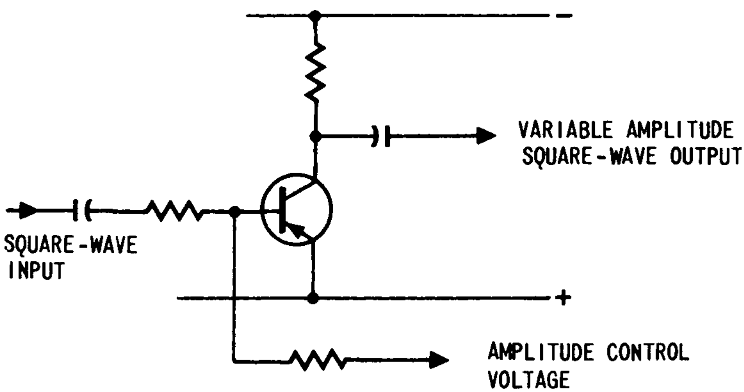


Fig. 4-21. It is possible, also, to vary the amplitude as shown in this circuit.

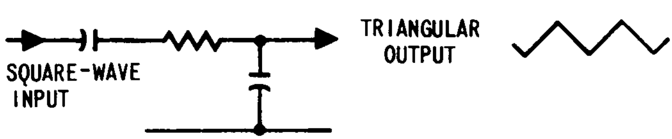


Fig. 4-22. This circuit will change a square wave to triangular.

Tone quality, or timbre, of a musical note is determined by its harmonic structure. A square wave contains a family of odd harmonics only. Making the wave asymmetrical introduces other harmonics; the more asymmetrical the waveform, the smaller the fundamental component and the higher in numerical order the concentration of harmonics extends. An asymmetrical wave can be achieved by changing one of the capacitors of the basic multivibrator, so the halves of the waveform are unequal. Changing one of the base resistor

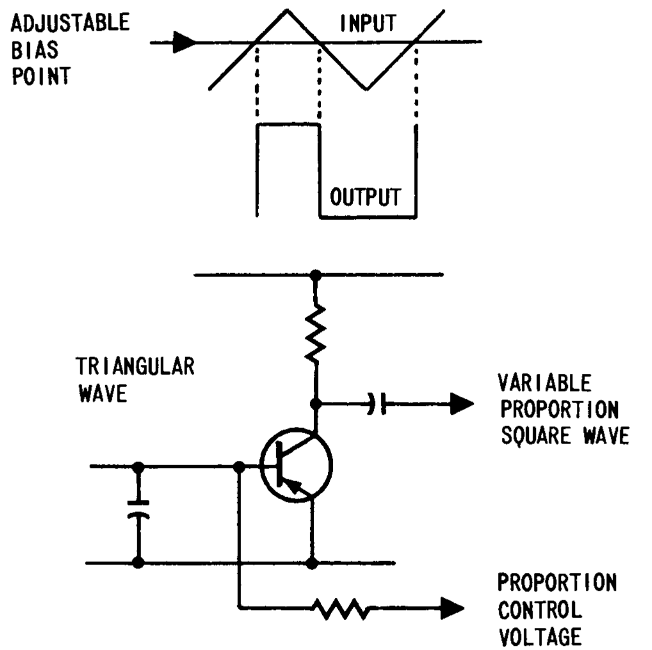


Fig. 4-23. The proportions of a square wave can be be varied electronically, without affecting its frequency.

values does the same thing. Applying the base resistors to different sources of voltage is one way of making the waveform, as well as the frequency, variable.

The simplest way to facilitate continuous variation of waveform proportions, without affecting fundamental frequency, is to produce a triangular wave by integrating the square form (Fig. 4-22) and then change the bias of a circuit that varies the point on the triangular waveform at which a saturated stage reconverts it to square (Fig. 4-23). Solid-state circuits to manipulate waveforms in any variety of ways are quite simple to devise. Also, formants, of the type used in electronic organs, can modify the sound by changing the waveform in other ways, using the harmonic content already present, either instead of the method just described or as an additional way of providing variation.

Chapter 5

Special Devices

While it is true that making the best of all the factors discussed in the previous chapters can make a tremendous difference to the overall performance of a system, there are some situations which still seem almost impossible. These are the situations that have prompted the design of many "special" devices that seem almost to work miracles.

A bad auditorium will set up a howl at so many frequencies that using a filter, a directional mike, or mikes, and everything we have mentioned, still leaves performance marginal. What else can be done? Or in another situation, background noise is so bad that the wanted sound seems irretrievable. Extra gain on a two-way communication system may cause howl problems, before either party can hear the other speak.

FREQUENCY SHIFTING

To cause a howl, the system builds up a standing-wave pattern at a particular frequency. The dominant frequency from the loudspeakers feeds back to the microphone through this standing-wave pattern, and the intensity builds up into a howl. We have discussed use of better speaker placement, microphones better in both frequency response and directivity, and selective reduction, by electronic means, of frequency response in the region where the howl starts. With all these precautions, a highly reflective building can still cause problems. Each of these steps may improve the situation but not quite enough. The system is still a problem, and a howl starts before a clearly audible sound is broadcast by the system.

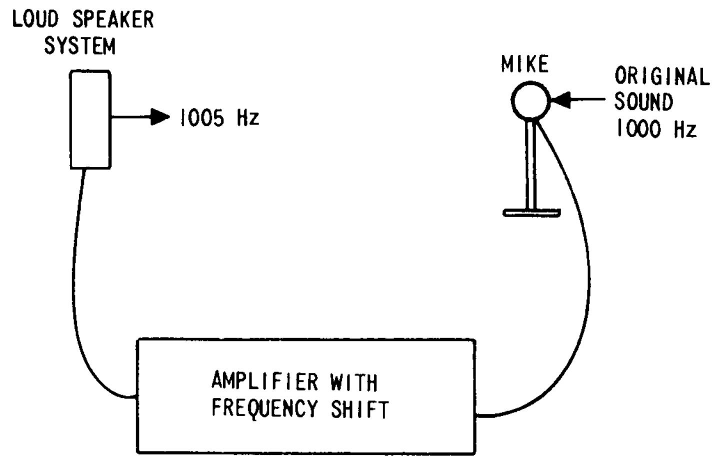


Fig. 5-1. A frequency shift helps to reduce the acoustic feedback tendency.

The build-up of a standing wave depends on the microphone picking up the same frequency broadcast by the speakers, amplifying it, and feeding it back to the speakers a little bit higher in intensity than before. One way to break this chain is to change the frequency a little bit in the electrical or electronic part of the system (Fig. 5-1). In this way, every frequency that comes from the loudspeakers is a little bit different, about 5 Hz, from the corresponding frequency fed into the microphone. This change enables a considerable increase in gain before a howl can get started.

Such a frequency change is achieved by applying the voice input to a frequency changer, which shifts the voice range of frequencies, from 20 to 20,000 Hz (that's the theoretical range, actually there is little at the extremes of this range) to, say 50,000 Hz plus these frequencies. Then the audio signal is shifted down again by another frequency changer, so that the final output yields frequencies just 5 Hz different from the respective original input frequencies.

This may seem very complicated, and it isn't simple. It requires two oscillators, which the frequency-changers combine with the signal in succession, whose frequencies differ by precisely 5 Hz. Oscillators used for this purpose must be stable. If they pulled together the shift would disappear and

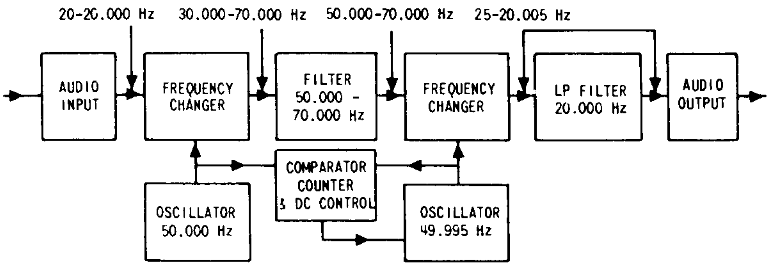


Fig. 5-2. Block diagram of a frequency-shifting circuit.

if they drifted apart it would become too great and become noticeable or objectionable. One way to prevent this is by means of counters feeding an automatic frequency control that sets one oscillator against the other so the difference holds at 5 Hz (Fig. 5-2). To complete the system, two sets of filters eliminate unwanted components after each frequency changer. On voice reproduction the presence of a 5-Hz frequency shift is barely noticeable, but on music the effect is impossible. Everything is completely out of tune. So where such a device is installed, it is imperative to provide means for bypassing it when music of any kind is being reproduced.

NOISE SUPPRESSION—HIGH LEVEL

Because noise often contains the same frequency elements as the signal, it is impossible to remove the noise by frequency-selective filtering, especially when noise level exceeds signal level. For all the similarity in frequency content, noise does have a character that is recognizably different from signal. This fact makes it possible for more sophisticated equipment to reduce noise in comparison to signal, although by the simpler analysis this would seem impossible.

The annoying part of noise consists of high intensity impulses that are non-repetitive and thus don't possess "tone," although they are contained within a frequency spectrum. So a device to remove or reduce them must be time- and level-discriminating, rather than frequency-discriminating. A whole variety of noise suppressors have been designed to work on this general principle (Fig. 5-3), which are capable of retrieving signal

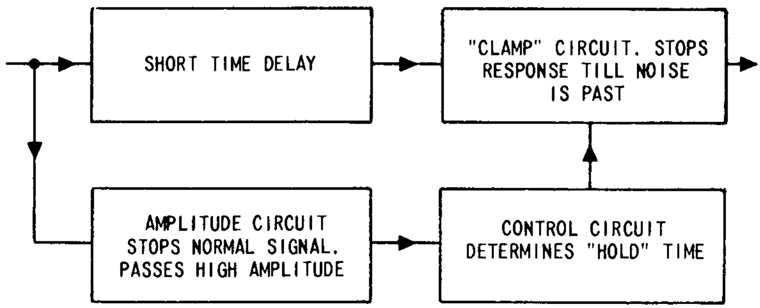


Fig. 5-3. This block diagram illustrates the basic concept of high-level noise suppression.

that previously seemed inaudible in the background noise. These circuits are able to detect the difference between the periodic type signals characteristic of wanted program, and the high-intensity impulse-type noise, which they suppress just for those instants, "holding" the audio signal while they do so.

Obviously, as programs contain some transients with characteristics similar to the noise being eliminated, some of these transients may be eliminated or drastically reduced by these circuits. But the "intelligence" which they do retrieve is 100% better than "nothing at all!" Some of these circuits restrict the high-frequency response range when they come into operation, thus limiting the byproducts of the noise. Others have a fixed high-frequency cut-off, compatible with the impulse period to which they become sensitive. Which suits a given application best depends on the type of noise present. The only way to be sure is to try out available types.

NOISE SUPPRESSION—LOW LEVEL

An older method of controlling noise is to change the dynamic range of the signal. This is effective where noise obscures only part of the signal—the lower level portions. The classic method of doing this is compression and expansion. Dynamic range is the level difference (in db) between the highest and lowest level signals in a program. The high-level limit is the power handling capacity of the system. The low-level limit is residual noise.

Every system has an available dynamic range. Power amplifiers may run 90 db or better. Preamplifiers are usually 60 db or better. And every program source has a dynamic range, between the highest and lowest level of component frequencies in the signal delivered. Noisy systems, with the exception of those where noise exceeds signal level, are classified as those where noise is not sufficiently below maximum, or highest level signal, to allow the lowest levels necessary for effective intelligence or appreciation of the program to be heard. The situation can be improved by reducing the dynamic range of the signal to match the dynamic range of the system at its minimum point.

COMPRESSION AND LIMITING

Compression must be achieved without unacceptable distortion of the signal waveform. What is "acceptable" depends on the nature of the program. For voice a progressive limitation that works on instantaneous signal can be quite acceptable (Fig. 5-4). But for music this is completely unacceptable. This technique distorts a sine wave of maximum (or even lesser) amplitude very considerably, and where any two notes are present simultaneously, considerable intermodulation will occur. This means compression of musical program must be achieved in some way that does not distort the waveform at any particular time. It must work like a manual gain control that intelligently turns up the quieter passages and turns the louder ones down.

To automatically achieve such control, sudden level changes must be anticipated and appropriate action taken as the level changes. As an apparently loud signal appears the gain must be reduced virtually instantaneously to avoid overload. As level drops following such a crescendo passage, the gain must gradually be turned up again to maintain lower level signals adequately above the background noise. Such a control requires carefully designed time constants, and a change of level that does not try to "follow" the amplitude variation or waveform of the lowest frequencies present in the signal. Of course, this will depend to some extent on the amount of compression to be applied. If a 60 db dynamic range must be

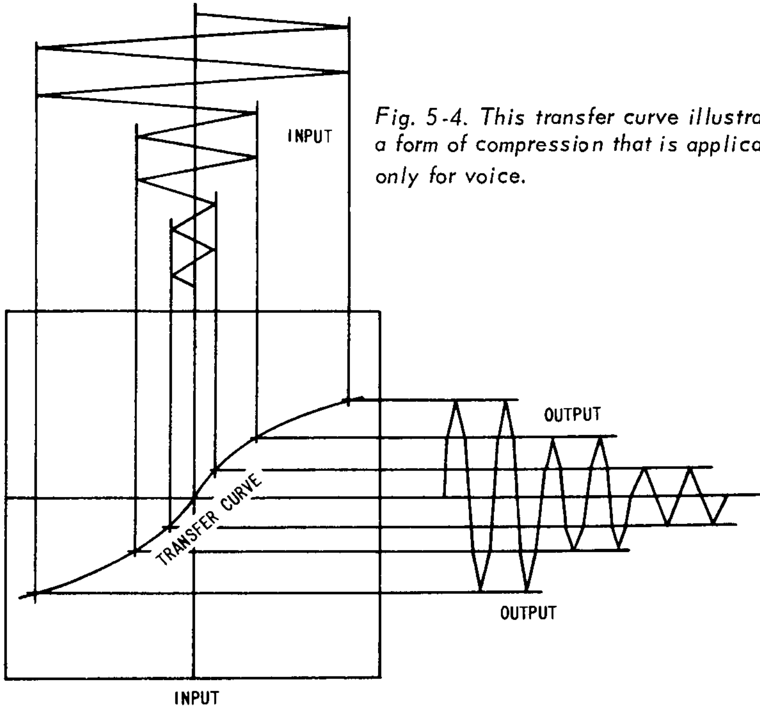


Fig. 5-4. This transfer curve illustrates a form of compression that is applicable only for voice.

compressed into 40 db, the problem is less rigorous than if the target is a dynamic range of 20 db.

Quite different from compression, and more effective for some purposes, is the action called limiting in which the peak output level is maintained as close to a specified maximum as possible. A typical limiter would compress 40 db of peak dynamic range (measuring only the peaks of the waveform) into about a 6 db variation of peak level, possibly less.

A compressor starts to reduce gain from the lowest level in the signal's input dynamic range, on up, so the variation in gain is proportional to the peak level at any moment in such a way that peak level is proportionately reduced. For example, a signal with peaks 60 db below maximum level would be increased, say 30 db. One with peaks 40 db below maximum would be raised 20 db. One with peaks 20 db below maximum level brought up 10 db, and so on (Fig. 5-5). The dynamic range is compressed (in this example) into half its original value. A limiter, on the other hand, would do nothing until peaks reach almost the maximum output level when the system is at maximum gain. Then the system would begin reducing gain so that any further increase in peak input level does not increase the peak output level materially (Fig. 5-6).

Some components serve a dual purpose, combining the functions of these two in different ways. The usual applies compression over the lower part of the dynamic range, and then adds limiting at the top end as a safeguard against overloading should the input dynamic range exceed the top level for which compression is designed (Fig. 5-7). The main distinction between the function of a compressor and a limiter circuit is the method of obtaining control voltage. For the compressor it may be obtained by isolating and rectifying (with appropriately designed high-speed "up" times and lower speed decay times) signal from either input or output (Fig. 5-8). The amplitude of the rectified output needs modifying so the adjustment effected in the gain control stage is as linear as possible.

For limiting the control voltage must be taken from the output, and a delay voltage must be used so that signal whose peaks do not reach that level do not get rectified. Only peaks exceeding this delay voltage produce a control voltage which then rapidly cuts back gain, so the peaks just reach the maximum allowed (Fig. 5-9).

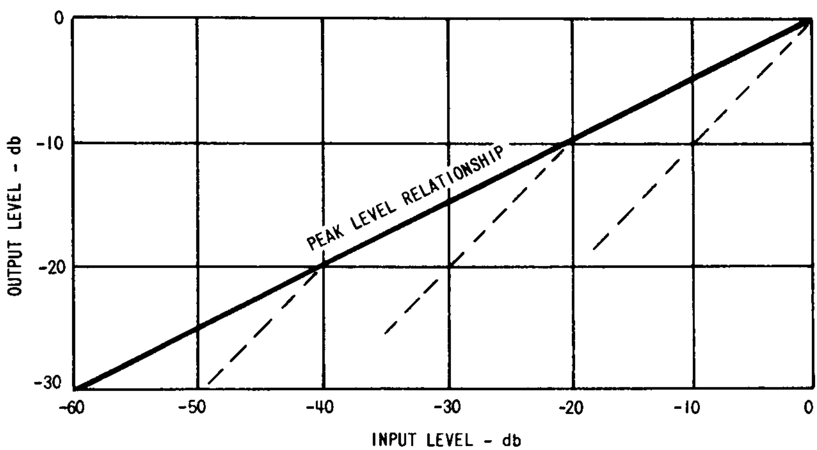


Fig. 5-5. In this graph the solid line represents the relationship between peak input and controlled peak output. The dashed lines indicate instantaneous input/output relationships, operating below specific peak levels.

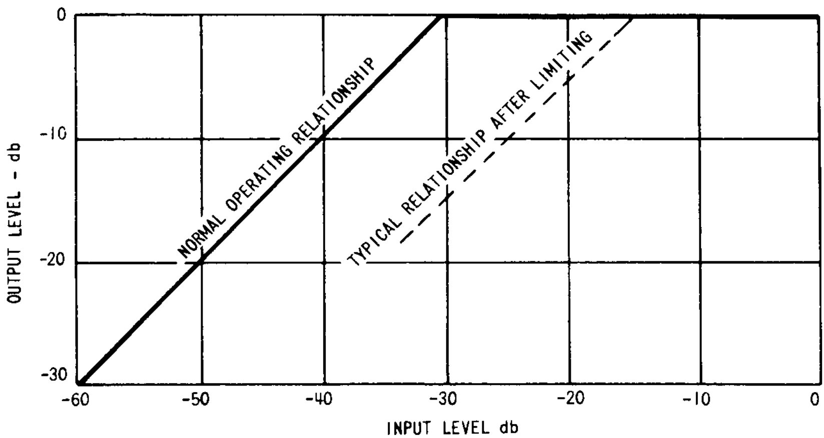


Fig. 5-6. These curves portray the function of "limiting."

A combined compressor-limiter needs two sources of control voltage: one without delay to achieve compression, and a larger one (with greater amplification) with offsetting delay voltage to achieve limiting. Where only a portion of a system imposes dynamic range limitations, such as a microwave radio link, it may be desirable to retrieve the original dynamic range, or some of it, after the signal passes through this restricting link (Fig. 5-10). This can be done with an expander, which just inverts the operation of a compressor.

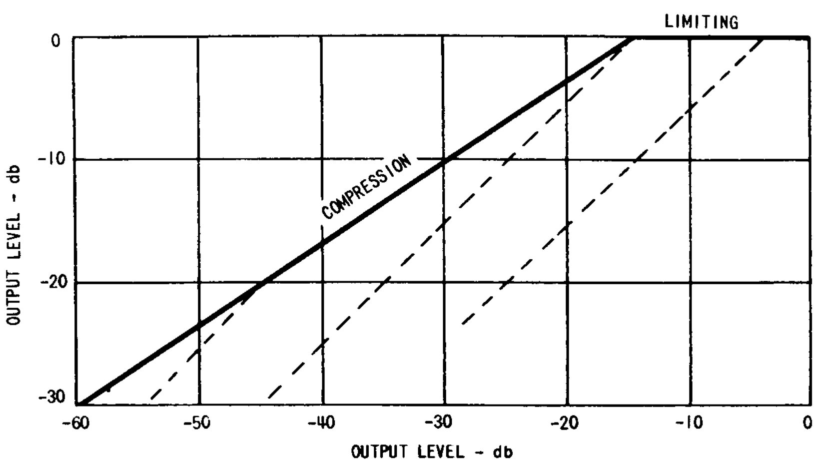


Fig. 5-7. Here, compression is combined with limiting.

With well-designed complementary compressor or expander systems, the output signal can be a close replica of the original input. The control voltage must be derived from the output of the compressor and the input of the expander, which have substantially the same program content and the same dynamic range. Only on transients, where a sudden change of gain is needed in both units, will there be a tendency to fail. And careful design can usually disguise this fact.

One improvement can almost eliminate detection of the fact that the system contains these units: inclusion of a code signal to control the expander from the exact same source as that controlling the compressor. This code signal is sub-audible, either infra or ultra sonic. It is separated at the receiving

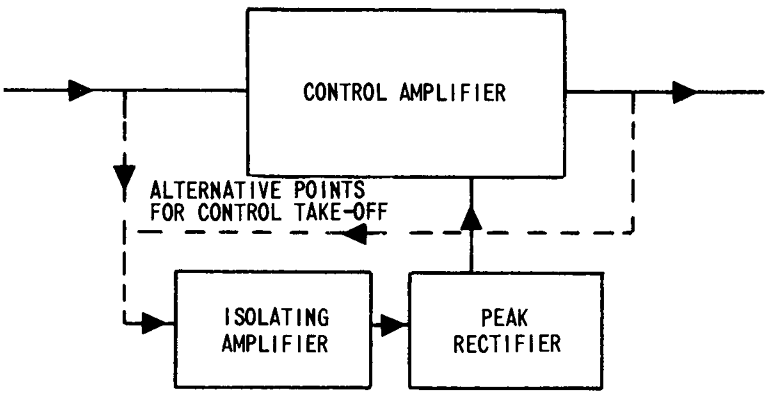


Fig. 5-8. Compressor block diagram. Dashed lines represent alternative points for deriving the control.

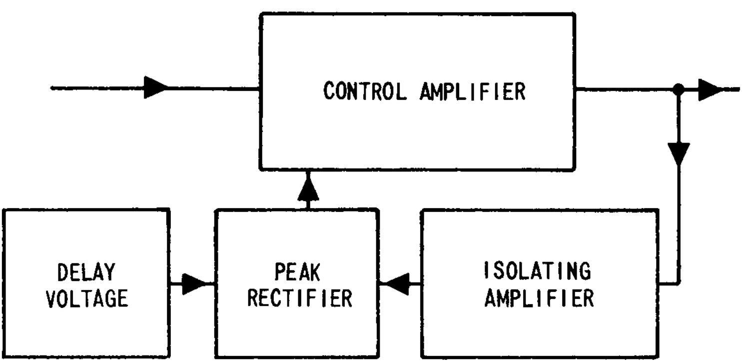


Fig. 5-9. Block diagram of limiter.

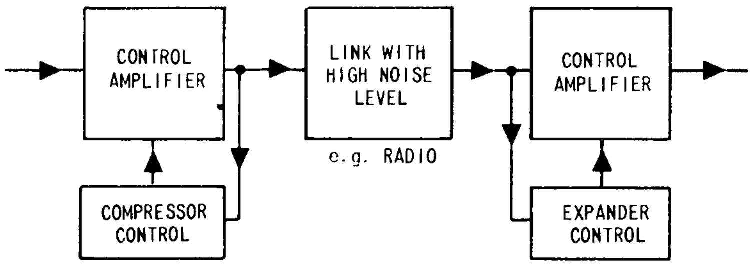


Fig. 5-10. Connected as indicated here, a complementary expander used with a compressor restores the original dynamic range.

end and used to control the gain of the expander directly (Fig. 5-11). The advantage of this addition is that peaks requiring gain reduction to avoid overload can be anticipated by using a delay, or by using manual control and providing the operator with some kind of cueing. Where an automatic control (with delay) is used the coding signal is derived from an undelayed signal. The signal passed to the system is delayed by a few milliseconds to allow time for the control to act before a high peak reaches the gain-controlled stage (Fig. 5-12). At the same time the coding signal is added to the output from this compressor, so the expander restores the original signal by changing its gain in a manner that exactly complements the compressor.

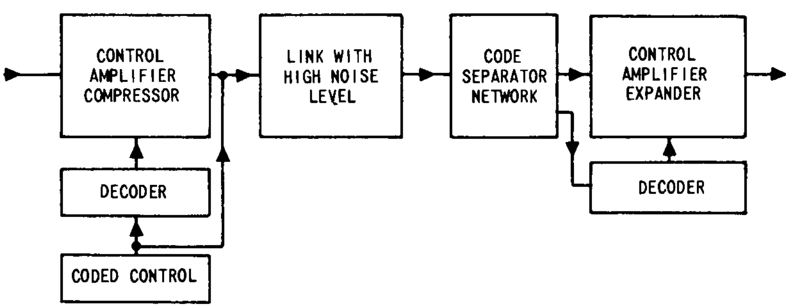


Fig. 5-11. As this block diagram suggests, coded controls improve complementary functions.

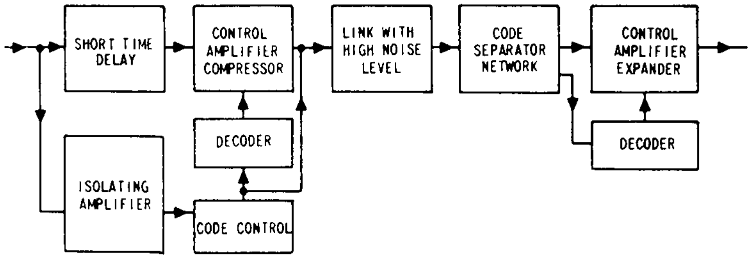


Fig. 5-12. Coded control in this block diagram is aided by a time delay in the main signal channel.

PRE-EMPHASIS

Neither the top nor bottom limits of dynamic range are as simple as it first appears. We have seen that noise can perform different roles at the low-level end. Overload at the high-level end is more than a simple magnitude factor. Normal signal contains most of its energy in the mid-range, below 1,000 Hz. Human hearing exhibits maximum sensitivity in the region from 2,000 to 3,000 Hz, with not so much reduction above this region as below it. This is why pre-emphasis can be useful in establishing an apparent margin above noise. But pre-emphasis can sometimes push the upper levels over the limit (Fig. 5-13). When the higher frequencies have been pre-emphasized, they may exceed maximum power level, although before pre-emphasis the limit may have been set by mid-range frequencies. This means that limiting, or compression, should be based on a signal obtained from a control derived after pre-emphasis is added. One thing this achieves is called "de-essing."

Where compression is used with de-essing, initial sibilants at the beginning of words tend to get over-emphasized. Imagine that I am describing this to you in a radio or television program that uses compression. I pause, then start speaking again, "Suppose ..." The actual energy level in that "s" sound is quite low, but quite audible. The energy level in the later part of the word is higher and will turn the gain down by actuating the compressor. So the effect is as if I spoke with an emphasized whistling through my teeth at the beginning of

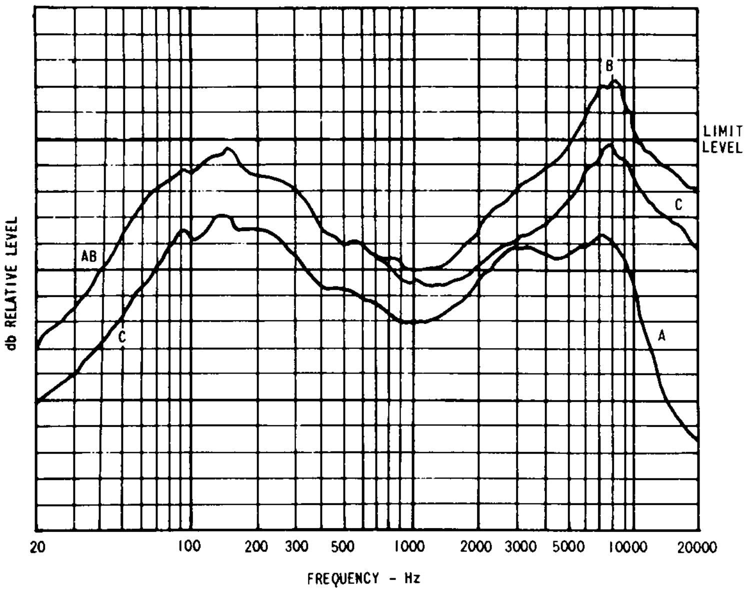


Fig. 5-13. These curves illustrate how pre-emphasis can invalidate compression: curve A represents the spectrum of a signal, without pre-emphasis, in which a frequency at about 150 Hz sets the limit; curve B, after pre-emphasis higher frequencies in this spectrum exceed the limit; curve C, when the pre-emphasized signal is compressed, the lower frequencies are reduced in level as well.

words that start with "s." The "s" of "suppose" comes out loud and emphatic, and the "uppose" part is turned down to a relaxed-sounding level. This effect can get very tiresome to a listener.

Actually, full de-essing may require more than normal pre-emphasis in the signal used for control. What it needs is an emphasis that raises the gain, approximately, to the frequency present in the "s" sound, in proportion to the amount by which human hearing is more sensitive to these sounds as compared with mid-range sounds. Another approach to limiting, where the peaks are too high when pre-emphasis is added, is employed by the Fairchild Conax. Whichever method of deriving control for compression or limiting is used, the result can sometimes sound unnatural. In music a cymbal clash is rich in such high-frequency components. What does the compressor do to such a sound?

Without pre-emphasis the cymbal-clash will undoubtedly cause breakup. If the program is applied to a phonograph recording (after pre-emphasis) the clash will result in a high-amplitude, high-frequency "burst" that will not be correctly transcribed on the disc, or if it is, it will be almost impossible for any pickup to recover without distortion. If the program is transmitted over FM, the high-frequency, high-amplitude component (after pre-emphasis) will produce sidebands far beyond FCC limits, so that the RF filters will be unable to keep the transmission within bounds. When such a transmission is demodulated following reception, this elimination of sidebands appears as distortion.

Where pre-emphasis is used in deriving the compression or limiting control, a cymbal clash can modulate other musical sounds unnaturally, in a manner similar, but opposite, to the "s" effect just described. When the cymbal clash "hits," the rest of the program gets "turned down" by it. The Fairchild Conax overcomes this problem effectively. First, it divides the low and high frequencies for separate treatment. The high frequencies are then pre-emphasized to determine relative peaks after this is added. Then the Conax clips, or attenuates, the upper end, but only enough to get the overall signal within the required envelope. After doing this, the high-frequency signal is restored to its de-emphasized state and recombined with the low frequencies.

This treatment alters the high-frequency end so little, without altering anything else, that the ear cannot detect any change. A demonstration of an entire system—from program input to transmitted or transcribed output—in which Conax is cut in and out (for A-B purposes), gives the impression that the only thing the Conax takes out is the distortion that occurs in transmission or record-playback. The difference can be seen on a scope, at points prior to the link in the chain where the distortion occurs without its use, but it cannot be heard.

GAIN-SHIFTING

For two-way, or multi-way, loud-to-loud communication, feedback can be a very real problem. If only one way is "open" at a time, there is no problem. But then it is impossible for the end who is not speaking to interrupt, as can be done on a normal telephone circuit. Voice-operated circuits have been in use for some time, but they have this short-coming.

If the voice-operated control energizes gain anytime there is an input to that circuit, people speaking at both ends at once will cause an acoustic howl, due to the two-way gain being too high, acoustically. If an electronic interlock is included so that whichever control increases gain first prevents the other from doing so (that is, until the first one's control releases) it is impossible for the other to interrupt.

Or it was, until the advent of gain - shifting, an innovation by Fairchild which keeps both circuits working at a sufficient level to be audible, but which may be a little too low for comfort in average surroundings. The voice control then increases gain at the transmitting end and reduces it on the receiving channel, so the overall acoustic gain is unchanged. Thus, in normal conversation, where only one person talks at a time, the person speaking has the lion's share of the gain, but the other channel is still open. If the other person wants to interrupt with a "Hold it there," he can do so. If both speak at the same intensity for that moment, gain will approximately equalize. If the interrupter speaks more loudly, then he will be received more loudly. The whole thing works very realistically and unobtrusively to make conversation more comfortable.

Another application where gain control is a little different occurs in installations where background level may change. In a sports arena, for example, crowd noises may necessitate a higher level. Without changing the level the commentator would not be heard when the crowd is noisy, or else he would be deafening when the crowd is quiet. Or perhaps the installation occupies a location under an airport approach. When a plane comes over, the system seems inaudible, and when the plane is gone, if the level was audible while the plane was passing, the system is now deafening. An automatic control, using the Fairchild "Ambicon," overcomes this by changing level according to a pre-arranged relationship with the background level, as picked up by microphones distributed around the area served.

REVERBERATION

The possibilities for adding to audio systems are almost infinite, especially now that solid-state circuits have miniaturized packaging beyond the wildest dreams of a decade or two ago, and have made possible some techniques that likewise couldn't have been done without them. In Chapter 3, we discuss a time delay, using a tape recorder with successive playback heads to eliminate effective reverberation from difficult installations. But time delays can equally well (and probably more often) be used for adding reverberation and kindred effects.

For artificial reverberation a simple tape record-playback system is not too effective. However many playback heads are used, the sound reproduced from them comes in too-definite "lumps." For this reason, either an acoustic echo chamber, or a system where multi-path allows a little more varied a mixture of feedback paths, is desirable.

Early attempts at providing artificial reverberation were very primitive compared to what can be done with today's sophisticated equipment. The spectrum can be divided into sections by means of fairly wide channel filters (octave filters serve well) which are then treated separately for the amount of reverberation added, plus time and amplitude composition, and then recombined into a composite signal. The variety of special equipment in which Fairchild Recording specializes enables almost anything to be done, according to the amount you are prepared to invest.

STEREO

Multitrack sound, either the simple two-track used for home and other domestic stereo, or the greater number used for theater and other presentations, or merely to have independent control of different portions of the original program when it is played back, merely involves the same kind of controls we have already discussed, but with a corresponding number of controls. One thing additional is needed—coordination between channels. It should be possible to limit, compress, or what-have-you, both or all the channels together, so the relative effect is preserved, as well as providing for a deliberate changing of the effect. Here again, the degree of sophistication possible is limited only by the imagination of the designer.

Some care is necessary to be sure one effect does not offset another. For example, if you want automatic compression or limiting to protect against overload, without interferring with other special effects, the control must use intelligence obtained from each channel, after you have put in the deliberate artificial effects, and apply the result to control both, or all channels, simultaneously.

REMOTE CONTROLS

These also fall within the realm of special devices. The problem with programs calling for remotes is that control is

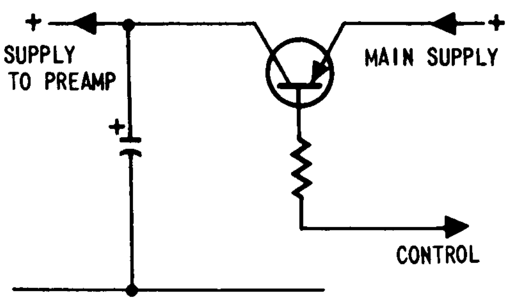


Fig. 5-14. Simple transistor "on-off switch" circuit controlled by voltage through the base resistor.

required from several places at different times. The precise method of operating remotes will vary, according to the purpose they are to serve. Usually each operator on the system needs to know what's going on, so he will be connected by a monitoring circuit, often in the form of an earphone, that enables him to hear the output of the system. Superimposed on this, or by separate channel, each operator also has communication with the headquarters point, or the coordinating center, so two-way communication can allow best use of program time to be utilized, based on what's going on and where. Coverage of political conventions is a typical example of such a system.

At one time, remote operation had to be manual, or else controlled by relays of the electromagnetic type, which were never exactly trouble-free, although much work went into improving their reliability. And although they might work reliably, they could get noisy, even if they weren't to begin with. Here again, solid state has come to the rescue. Transistors operate extremely effectively as relays, and silently, too. Also, the light control developed by Fairchild can be applied in various ways.

One simple way to make your own is to use a transistor as

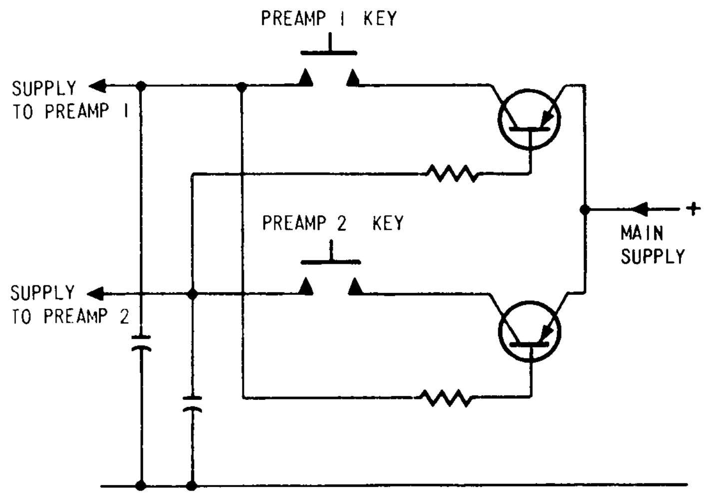


Fig. 5-15. Transistor switch circuits are quite suitable for electronic interlock purposes.

a voltage-supply control in the gain stages of preamplification (Fig. 5-14). Transistors do not make good variable gain stages, but they can make good variable-resistance attenuators, as shown in Fig. 4-12. However, turning their supply voltage on or off with a noiseless switch is a convenient way of cutting different sources of program in and out rapidly, and by careful design, silently. For example, if two preamplifiers feed the same common channel, and they are energized by an electrical contact at a local point of origin, an electrical interlock can be made that will "hold" the circuit for whichever preamplifier happens to be working. That way, the other one cannot come on until the first is finished (Fig. 5-15).

Applying supply voltage to either amplifier destroys the bias on the series-feed stage for the other, so it cuts off. Then, even if the button (or other manual control device) that should apply voltage to the second preamplifier is closed, nothing happens because the feed transistor to that circuit is nonconducting until the first circuit loses its voltage, again turning the feed transistor on. When neither preamplifier is switched on, both feed transistors are held conducting, so that either can be switched on. But as soon as one of them is switched on, the other is switched off by the transistor. The possibility of each killing the other is prevented by taking the feed point for each bias point from the output side of the other's control. One of them must go on to cut the other off.

INTEGRATING THE SYSTEM

Having all the parts to do a job is not a guarantee of a good or properly-working system. In the course of discussing features, we have suggested where facilities can and cannot be used. But all these units must be inserted at the correct operating level and impedance. Chapter 1 emphasized this, but the mistake of overlooking it happens so often that it cannot be overemphasized. Also, facilities should be applied as closely as possible in association with the function to be served. For example, limiting should be added only immediately before the link—transmission or recording—where the limit must be imposed.

On the hardware to put a system together, we could write another book. At one time, everything was done with patchcords—and what a tangle one could make! Now solid-state switching is so much easier to accomplish that push-button consoles are the thing, using a multifaration of the principle illustrated at Fig. 5-15. With indicator lights built in, the operator can see at a glance what is operating and set up something different in a flash by touching a few buttons.

Chapter 6

The Complete System

From the technical point of view, there are various approaches to achieving what the "hi-fi" customer wants. But there is also a difference in customers, to which we should give some attention. The customers most likely to read this book are interested not only in listening to good reproduced programs, but also to at least the superficialities of the electronics involved. Other customers just want to enjoy the best possible reproduction; they're content to leave the ways and means in the hands of people they hope are experts.

ALTERNATIVE POWER PHILOSOPHIES

From the technical viewpoint these alternatives represent opposing approaches to the problem. One, historically the earlier, uses loudspeakers of the highest possible efficiency and can thus use conservatively rated amplifiers to feed them. If you used a 100-watt amplifier to feed a speaker system that would fill a good-sized auditorium, much less a living room, on about 5 to 10 watts, you may damage the speaker, and very likely someone's eardrums!

The alternative approach was really in response to a certain kind of demand: To achieve high-efficiency with uniform response at the lower audio frequencies, a speaker system has to be large so that it may efficiently develop the long wavelengths of sound involved. We'll discuss this in more detail in Chapter 9, but here we must recognize it as a fact of life. Since so many people live in apartments, trailers, and other dwelling places of limited space, they encounter some diffi-

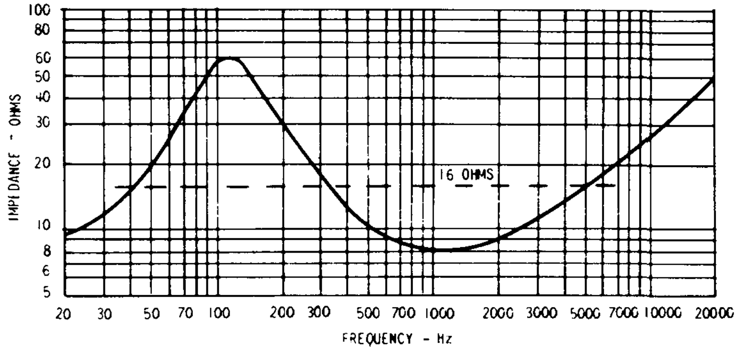


Fig. 6-1. Impedance curve of a typical loudspeaker that may be rated as either an 8-ohm or 16-ohm impedance. See text.

culty in achieving a high degree of fidelity in their sound reproduction. This led to the development of the "bookshelf" type speaker, which is essentially inefficient compared to the type used in the first approach. So this change in philosophy necessitated high-power amplifiers to achieve an equivalent sound level in the room.

Among the enthusiasts you will undoubtedly encounter, sooner or later, arguments about which approach is best, or even whether one or the other is even valid! The protagonists of each approach can become so extreme in their view that they deprecate the other as definitely, and in principle, inferior. Actually, as we shall develop in more detail in Chapter 9, nature has a certain logic that fits things to the sizes for which they are best suited, so that small speakers do a better job in rooms where space is at a premium, and large speakers are more effective in rooms that can comfortably accommodate them physically. So such arguments, treated logically, would really reduce to opposing declarations, one asserting that a small room, and another that a large room, is better to live in! Obviously this is a matter of personal choice.

POWER MARGIN—MATCHING

The general nature of loudspeaker impedance is discussed in Chapter 3, with reference to Fig. 3-1. The output of an amplifier is specified with reference to a constant resistance load, a value for which the amplifier output is designed. The fact that a loudspeaker's impedance is not a constant resistance leads to the first problem in matching.

Then there is the related question of required power margin. According to standard specifications a loudspeaker rated to handle 10 watts should be capable of handling a 10-watt signal at any single frequency within the range of frequencies specified as its response range. This is the most difficult test for a speaker to meet. Usually, it will handle 10 watts of almost any typical composite program material more readily than it will handle 10 watts at each and every individual frequency within its response range. Conversely, if it will handle 10 watts at any single frequency, it will handle the equivalent of considerably more than 10 watts of program signal.

It should be mentioned that 10 watts, as defined for this purpose, does not mean actual measured electrical power. As measured at any single frequency, the 10-watt level is determined by measuring output voltage across a load rated at its designated impedance. Thus, if Fig. 6-1 represents the impedance curve of a speaker rated at 8 ohms (which it would be according to standard specifications), 10 watts at 1,000 Hz would be just under 9 volts RMS. For the purposes of definition, 9 volts at any other frequency would rate at 10 watts, although at about 110 Hz, where the impedance is 60 ohms, the actual power would be only about 1.5 watts. If the same speaker were rated at 16 ohms impedance, then at about 45, 350, and 5,200 Hz the VA would be 10 with 12.6 volts RMS. At 1,000 Hz a voltage of 12.6 RMS would represent a power of 20 watts, while at 110 Hz it would represent a power of about 2.7 watts.

The discrepancy in actual power represented by uniform voltage is compensated for by changes in the unit's efficiency, which means the acoustic power output will be much nearer the same at all these frequencies. But the matter with which we are concerned in this Chapter is matching the amplifier, or the load that such speakers actually provide for an amplifier. The construction lines drawn on Fig. 6-2 represent conditions related to rating the speaker whose impedance is plotted in Fig. 6-1 as a 16-ohm speaker.

Line AOB represents a resistive 16-ohm load line drawn on the Class-B characteristics of two output transistors or tubes,

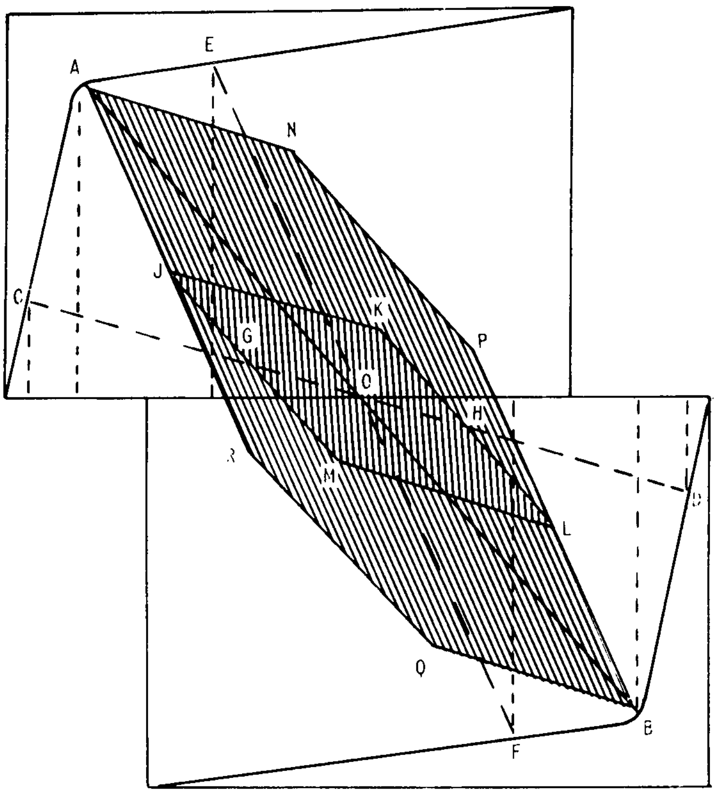


Fig. 6-2. Load-line construction on overall Class-B output characteristics to illustrate the effect of signals at different impedances and frequencies, related to the impedance curve of Fig. 6-1.

with the quiescent operating point O. As the speaker's impedance is reactive at frequencies such as 45, 350, and 5,200 Hz, where the value is 16 ohms the load would be an ellipse of the same general slope as the resistive line AOB. Where the speaker's impedance is resistive, which is at 110 and 1,000 Hz (approximately), the dashed-line load lines would apply. Line COD represents the load line at 110 Hz, and line EOF the one at 1,000 Hz. Notice that line COD represents a slightly greater voltage swing (dashed line projections on the horizontal axis) than AOB, but that line EOF represents about half the voltage swing. So, using single sinusoidal waveforms, the amplifier would deliver rated maximum output (in VA) at 45, 350, and 5,200 Hz. At 110 Hz it would deliver slightly more than rated voltage, which means only 2.7 watts into the actual impedance, and at 1,000 Hz it would deliver about half the rated output volts into half the rated impedance, or about half the rated watts, actual.

Using a speaker equivalent to a typical 8-ohm type to match the amplifier, it will deliver maximum rated power at 1,000 Hz, a slightly larger voltage, but into twice the impedance, thus slightly more than half the rated VA, at 45, 350, and 5,200 Hz, and slightly more than 1.5 watts at 110 Hz. As the improving efficiency in the vicinity of resonance makes up for the loss of electrical power, this seems the better arrangement, based on a sequence of tests at single frequencies.

However, a composite program changes the picture, and the remaining construction of Fig. 6-2 illustrates this. Here the partial load-line GOH represents a signal component at 110 Hz, about one-third maximum signal voltage at this frequency. Superimposed on this is a signal at, say 350 Hz, where the impedance is of nominal value. This opens out the load movement by the shaded area, parallel to load line AOB, resulting in the rectangle JKLM. Finally, a third signal at 1,000 Hz is imposed on this, necessitating a further excursion from the edges of this rectangle, parallel to EOF, and making the whole load area ANPLBQRJ.

Notice that this is not unlike the shape of an ellipse of the general slope of AOB, although it is made up of three frequencies working at once. This means the amplifier will deliver close to its maximum power at this composite signal of three representative frequencies. On the other hand, if eight ohms were matched to produce load line AOB, the whole shaded area would be halved in height, and the effective power output of the amplifier would be halved on the composite signal.

RELATIONSHIP BETWEEN PEAK AND AVERAGE POWER

Another aspect of the power-margin question concerns the relationship between peak and average power in program signal. This problem arises because amplifiers have to handle peak voltage (or current) rather than peak power. As instantaneous power is proportional to the square of instantaneous voltage or current, no direct relationship between the quantities exists. Musical tones are not simple sine waves. And programs consist of many musical tones put together. But to simplify the consideration, assume each musical tone is sinusoidal and that the program contains as many musical tones as there are instruments in the group. Thus, we might compare a five-instrument combo with a 50-piece orchestra. Assume that each instrument contributes the same energy level, say half a watt peak, or quarter of a watt average. Across an 8-ohm speaker reproducing this sound, this would be 2 volts per instrument.

As each instrument generates a different frequency, there will be moments when each instrument's waveform adds to each other instrument's waveform, peak adding to peak. And occasionally, say once a second, all the peaks will add at the same moment. The amplifier has to be able to handle this peak, although the average level is the sum of the average powers. For the five-piece combo, the maximum peak is five times two, or 10 volts. This is a peak power of 12.5 watts, or an amplifier with a power rating of 6.25 watts is needed to handle it. The total average power is five times one-fourth watt, or 1.25 watts.

For the 50-piece orchestra the maximum peak is 50 times two, or 100 volts. This is a peak power of 1,250 watts, requiring an amplifier with a power rating of 625 watts. But the total average power is only 50 times one-fourth watt, or 12.5 watts. This means that, making these simplifying assumptions, an amplifier to deliver 1.25 watts of combo music needs a rating of 6.25 watts, while an amplifier to deliver 12.5 watts of 50-piece orchestra music needs a rating of 625 watts!

Suppose we decide that a comfortable listening level corresponds with five watts average electrical power converted into acoustical energy by the loudspeaker. Now we will raise the combo level and reduce the orchestra level, as compared with the original performance. Now the power-handling capacity of the amplifier needs to be 25 watts and 250 watts, respectively.

In a Class B amplifier (which most of the larger amplifiers are) the current drain from the rectified supply depends not on the peak power but on the average power. A momentary peak merely takes a little charge out of the supply reservoir capacitor. A continued average power increases the current drain permanently.

Realizing that an amplifier's ability to handle momentary power peaks is the factor that controls how loud a program can be reproduced, more so than its ability to produce sustained power of single sinusoidal form, a few decades ago some amplifier manufacturers started to rate their amplifiers by peak power, in addition to quoting continuous or sustained output power as it always had been measured. To do this they artificially maintained the supply voltages at their quiescent values long enough to get readings at the power levels measured. Suppose that a quiescent supply voltage is 250 volts. But on sustained maximum output, it drops to 230 volts, and the power measured is 45 watts, average (the usual for a sinusoidal waveform).

Because the waveform is sinusoidal, the peak power is just double this, or 90 watts. But this isn't the waveform ceiling of such an amplifier, because with a musical signal the supply voltage would not thus be loaded down from 250 to 230 volts. So a test is made holding the supply voltage up at 250 volts, and now the average power reads 57.5 watts, so the peak would be 115 watts.

Based on these tests, the manufacturer of those times would rate his amplifier: sustained power 45 watts, peak power 115 watts. This involved a little more sophistication in measurement than some other manufacturers were ready to embark upon, unless a standard required it. So the latter (or some of them) started merely doubling the ratings: if they had previously rated their amplifier at 50 watts, they called it 100 watts peak.

If this reflected laziness, or an unwillingness to invest in the necessary extra equipment, at least it was valid. An average power of sinusoidal form must be twice that peak power, even though such doubling does not reflect the ability of the amplifier to handle momentary peaks higher than those in a sustained sinusoidal waveform. So other manufacturers, who had not studied musical waveforms so much (or the behavior of amplifiers when handling them) preferred to call an output of single sinusoidal waveform, maintained "solid power" or "honest watts" with the implication that "peak watts" weren't quite honest, perhaps!

While the manufacturers who initiated the peak-watts measurement were in no way dishonest, and only hoped to introduce a more meaningful method of measurement, the lay public is not well enough educated to appreciate this, and the implication of dishonesty was enough to persuade them to discontinue the practice.

All this happened before the Institute of High Fidelity Manufacturers came into being. When this group was formed, with the thought of introducing honest practices and meaningful standards of measurement, they realized that the simple average power on a sustained signal still did not relate adequately to an amplifier's performance on program. Unwilling to renew the old dispute about honesty, they brought back the same method, but with a different manner of specifying the result. For the peak operating condition they specify the average power, thus the amplifier described earlier would be specified as having a sustained power output rating of 45 watts and a music power output rating of 57.5 watts.

While the new combination of figures is more informative in relative terms—it shows how much more music power than sustained power the amplifier will give—it is less meaningful in absolute terms because the average value of a peak power rating is academically an anomaly. But electrical power ratings do not bear any definite relationship to loudness, or acoustic power, and the IHFM ratings represent a real step forward.

FREQUENCY DIVISION OF POWER

The foregoing discussion covers most of the aspects of power-rating problems, as related to delivering electrical power to a speaker system regarded as a single entity. But in the quest for better quality reproduction, the trend moved toward multiway speaker systems, with individual units operating within specific frequency ranges. First, two-way systems and then three- or more -way systems came along. To feed them, electrical crossovers were provided with the assembly that divides the output power by frequency, so the correct frequencies get to each loudspeaker unit. But this can aggravate the power-margin problem illustrated by the comparison between a combo and an orchestra.

Suppose we consider a 3-way system, using a big woofer to handle bass frequencies up to 150 Hz, a mid-range speaker to handle frequencies from 150 Hz to 2,000 Hz, and a tweeter to handle those from 2,000 Hz up. Assume the maximum capacity of the woofer is 16 watts (32 watts peak) the mid-range, 4 watts (8 watts peak), and the tweeter 1 watt (2 watts peak). Using an 8-ohm impedance, those peak ratings correspond with peak voltages of 16, 8, and 4 respectively. This means that the amplifier must handle a peak of 16 + 8 + 4 = 28 volts peak before these voltages are separated into the respective circuits. And 28 volts peak represents 98 watts peak, or 49 watts average power.

So we need a 50-watt amplifier (using a round figure) to deliver 16 watts to the woofer, 4 watts to the mid-range, and 1 watt to the tweeter. The problem is this: Having installed this amplifier size, there is nothing to prevent 50 watts of

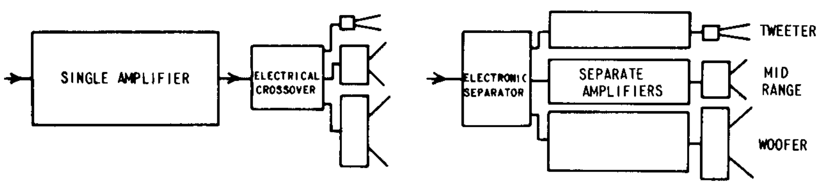


Fig. 6-3. These block diagrams illustrate the difference between an electrical crossover and electronic channel separators.

average power being delivered to any one of those units, which means that any one of them can be over-run, even the woofer, and the tweeter very much so. And under these circumstances, the combo is more likely to cause trouble than the orchestra, because it can concentrate more energy into fewer frequencies, if the gain is turned up too high.

The remedy is separate amplification for the individual units, a practice which led to the introduction of bi-amplification and tri-amplification. The difference is illustrated at Fig. 6-3. Now the woofer can use, say a 20-watt amplifier, the mid-range a 5-watt amplifier, and the tweeter possibly another 5-watt amplifier, as a 1-watt amplifier may not be readily available. The risk of damage is greatly reduced.

Moreover, the possibility of intermodulation distortion is also greatly reduced, because the frequency combinations

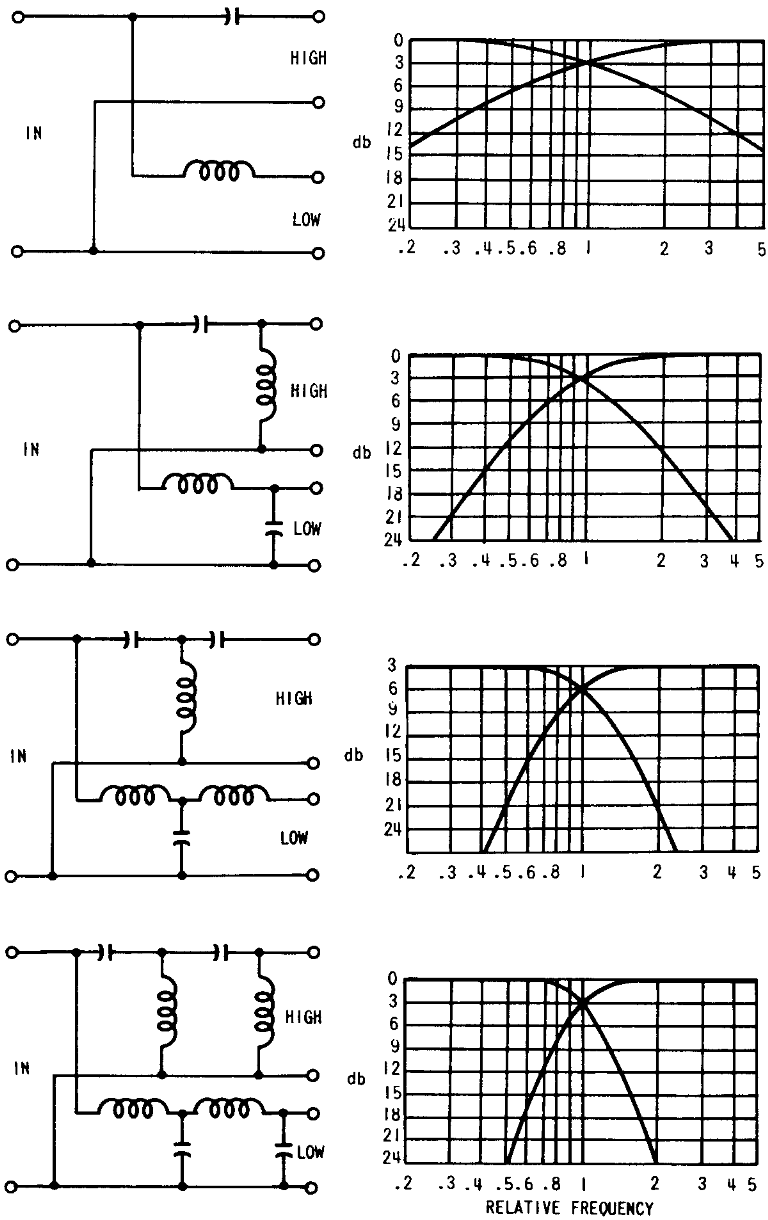


Fig. 6-4. Typical electrical crossover circuits, using from one- to four-element pairs, with the response produced when the networks have the correct values for all elements and termination.

that could cause trouble in the speaker system are separated earlier and thus cannot get into the wrong part of the system.

One thing has lagged, though, at the time of this writing—electronic separator units that provide separation comparable to electrical crossovers. In electrical crossovers, terminated by the impedances for which they are designed, the slope of the transition from one output element to the other depends on the number of reactance elements in each network (Fig. 6-4).

A single reactance in the feed to each unit results in a 3 db/octave slope at the crossover frequency, and a final slope of 6 db/octave. Two reactances in each result in a 6 db/octave slope at the crossover frequency, and an ultimate slope of 12 db/octave. Three reactances in each produce a 9 db/octave slope at crossover, and an 18 db/octave ultimate. And so on. The sharper the cut-off the more critical the values, and the more essential it is to have correct termination. Also, more phase shift occurs between different frequencies in the neighborhood of crossover, which may make the performance on transients less desirable than with the simpler, less rapid rates of transition.

A fact that seems to escape attention in the "electronic" frequency separator is that achieving the correctly shaped roll-off is more than having the necessary number of reactance elements. In the electrical circuits of Fig. 6-4 the values are chosen so the elements interact to produce the correct response shape. When successive RC pairs are cascaded (Fig. 6-5) to make a steeper ultimate roll-off, what they really do is multiply the effect of a single pair. If the turnover frequency for them all is that designated as crossover, the loss in each output at crossover frequency is 3 db multiplied by the number of element pairs used.

The correct shaping should maintain constant total power transfer throughout the frequency range, which means each should have 3 db loss at crossover. Some have shifted all turnover points, so that the points where the total loss is 3 db coincide, rather than the points where each pair contributes 3 db. This is not satisfactory (Fig. 6-6) because the response of such an arrangement, up to well beyond crossover, differs very little from the single RC network. The only result of adding more pairs of elements, when treated this way, is to accentuate loss in the unused output at frequencies an octave or more from crossover.

The only successful way to achieve electronic separation with the proper crossover response is to utilize feedback to modify the shape of the combined response. If two identical pairs, without interaction, have 6 db feedback applied overall, the turnover frequency is extended by root 2, and the 6

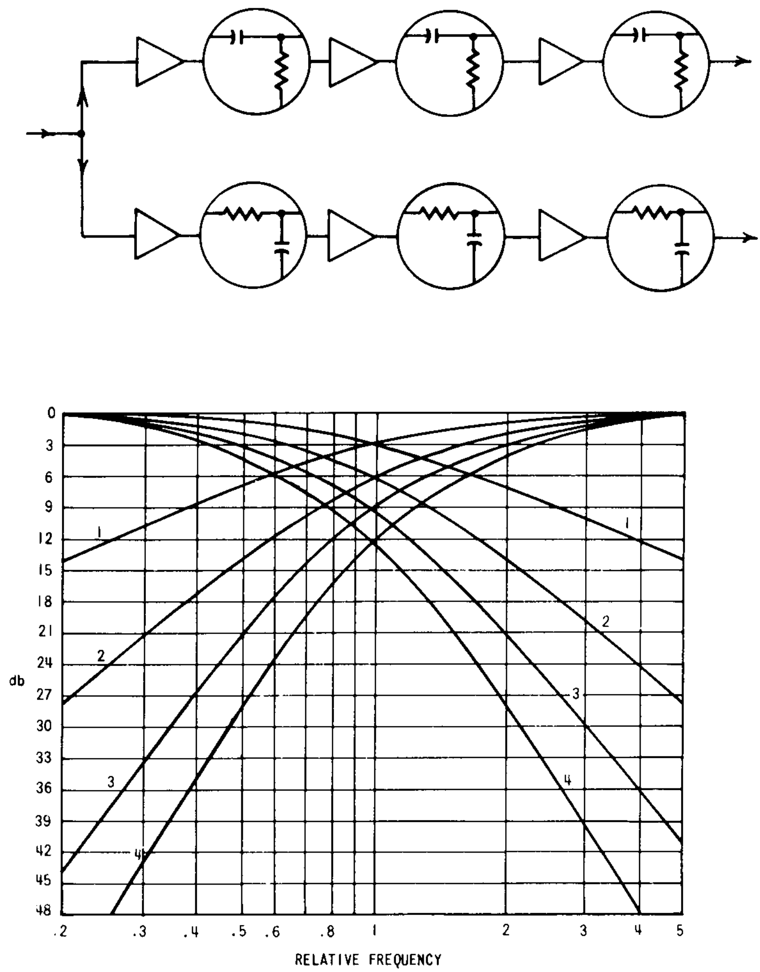


Fig. 6-5. Shown here is the affect of additional successive roll-offs on total response, without interaction or the use of feedback.

db/octave slope point is raised from 6 db to 3 db loss (Fig. 6-7). This is utilized by selecting pairs whose turnover acts at a frequency that is a factor of root 2 inside each section's passband. Then the feedback changes response to the correct shape and crossover frequency. Fig. 6-8 shows the relevant response components, with the block schematic.

For an ultimate slope of 18 db/octave, or 9 db/octave at crossover, two roll-off pairs inside a feedback loop are combined with one outside (Fig. 6-9). The turnover points of the roll-offs inside the feedback loop are an octave inside the passband, while the external one is at crossover frequency. For an ultimate slope of 24 db/octave, or 12 db/octave at crossover, the two roll-off pairs inside the feedback loop have their turnovers an octave and a half inside the passband, and 18 db of feedback is used. Then the external pair of turnovers are at crossover, and the overall response comes out right (Fig. 6-10).

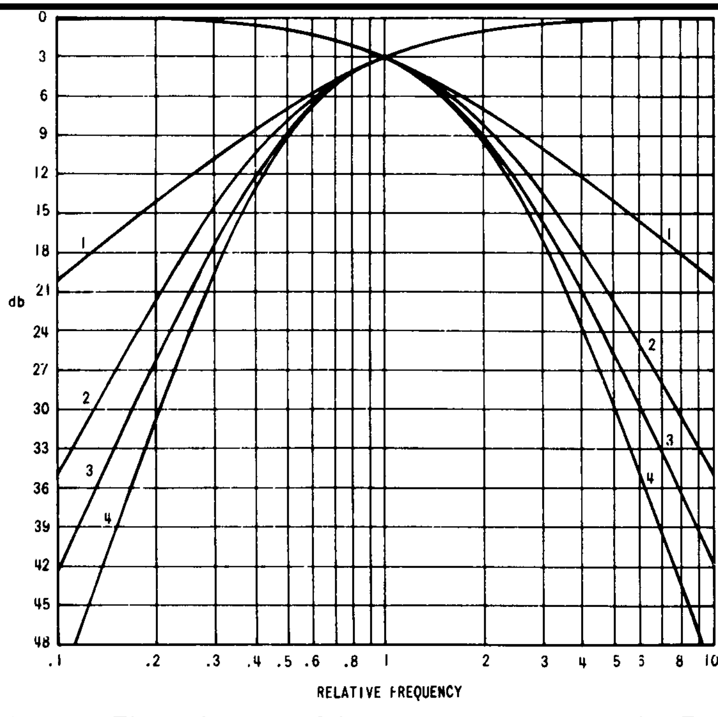


Fig. 6-6. The performance of the arrangements represented in Fig. 6-5 can be changed as shown here by sliding the response together.

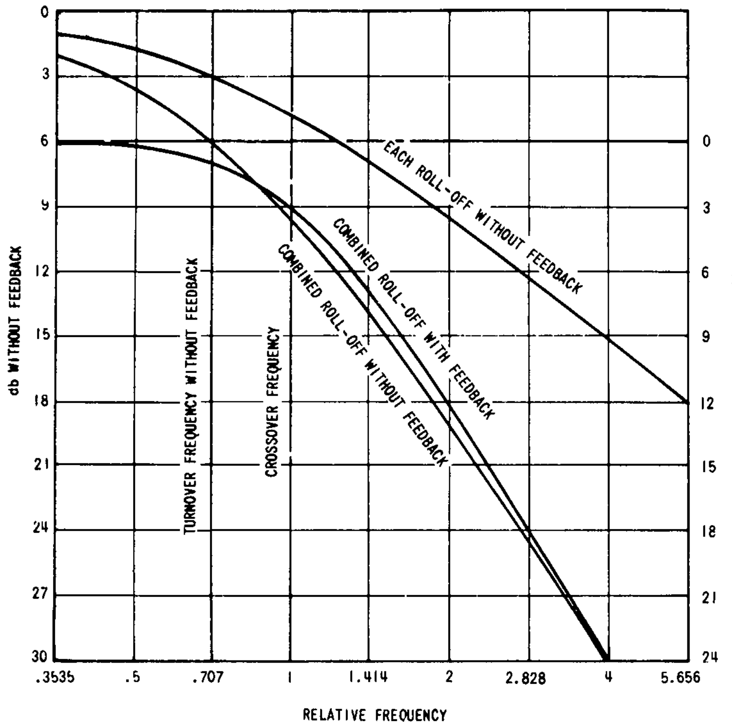


Fig. 6-7. These curves demonstrate how feedback modifies the response of two single-reactance roll-offs to get the required sharpness for a 12 db/octave crossover, which has a slope of 6 db/octave at the crossover frequency.

For multiway dividing networks, at least one of the units requires a bandpass. This can be achieved conveniently by combining component values, allowing for the effect of the elements acting "at the other end" in each case. As the effect is mutual, reactance elements are modified by complementary factors in each case. At this date, there is no commercial product available utilizing these principles, although many have built them for themselves, from first principles.

THE AUDIO CENTER

This is the core of any home hi-fi or stereo system. Basically, it is a simple unit with many functions; however, many different arrangements are used to appeal to different users

—very much like the automobile industry! The essential functions are to provide for the selection of program source: radio, phono, tape; to provide equalization where the source does not provide it; to provide some degree of tone compensation, thus optimizing the sound reproduction of any program type in the actual listening environment.

The object is invariably to combine flexibility with simplicity of operation. Flexibility requirements tend toward multifarate controls. Simplicity demands keeping their number to a minimum to avoid confusion. For example, some makes use separate controls for left and right stereo. Others gang both controls on one shaft so a single knob turns both, and then provide a balance control to adjust for any differences between channels. The latter method is generally regarded as best, combining flexibility with simplicity, because level can be changed by turning only one knob.

PACKAGE vs COMPONENT

A decade or two ago, these categories represented completely separate segments of the industry and often members of one would try to pretend the other didn't exist. Since then there has been some blending, to the mutual advantage of both.

The term "package" applies to a hi-fi or stereo system bought as a single entity from one manufacturer. Often the whole system would be contained in one package, usually a large console. But even where the speakers came separately, the set was bought as an entity.

The term "component" applies to the number of separate units that make up a system, which could be: radio tuner, tape deck, phono turntable or record changer, phono pickup, preamplifier or audio center, and loudspeakers. The enthusiast could put together his own system by buying components made by a diversity of component manufacturers.

In those earlier days the package manufacturers put together a complete system which the component manufacturers regarded as "cheap and dirty" and not worth dignifying with the title "high fidelity." The package manufacturers persisted in calling their products "Hi-Fi," so in those days the terms "high fidelity" and "Hi-Fi" had quite different meanings.

Things have changed since then. The package manufacturers have taken many leaves out of the component manufacturers' book to improve their products. And really, the package manufacturer does have one advantage: he designs the entire system so that the components "go together," rather than designing each without reference to any particular companion unit. The thing that most often reflected unfavorably was that

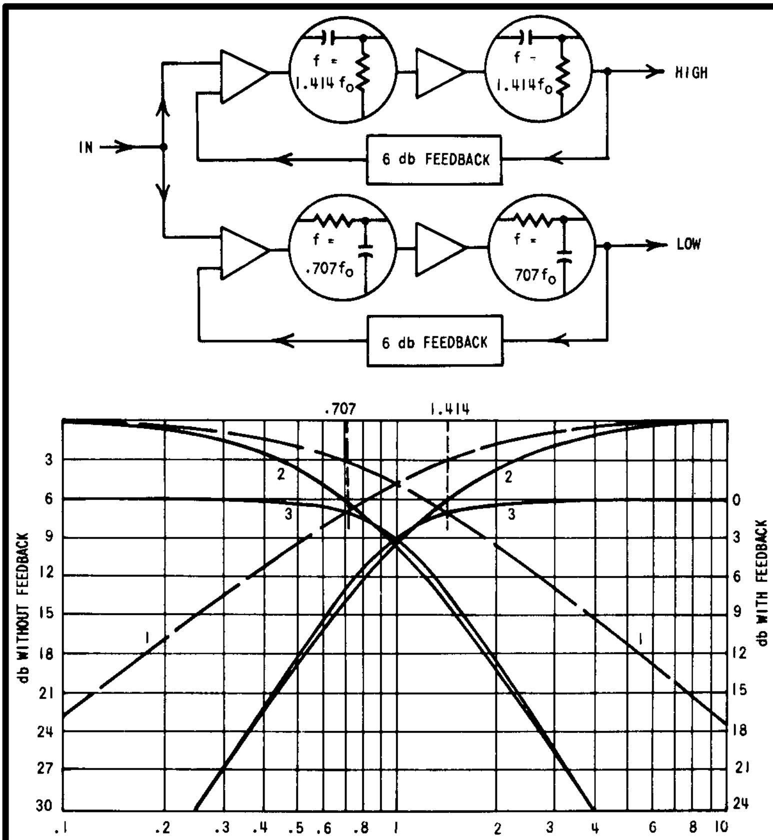


Fig. 6-8. Shown here is the method of obtaining a correctly designed electronic crossover with a 12 db/octave ultimate slope and a 6 db/ octave at the crossover frequency. Top, the circuit arrangement; bottom, how the response is built up. Curves 1, the response of each roll-off within the feedback loop; curves 2, the response of both roll-offs before feedback is added; curves 3, the final response formed by adding the feedback.

the package product sold at a price at which real high fidelity was not possible.

Now, component manufacturers have gone into the better class package business. They recognized that often the advantage of buying the best in component category was negated because the components thus selected didn't match, in one way or another. So the trend, even among component manufacturers, was toward an integrated system, which means a system in which the parts are designed to "go together," rather than as separate, unrelated entities.

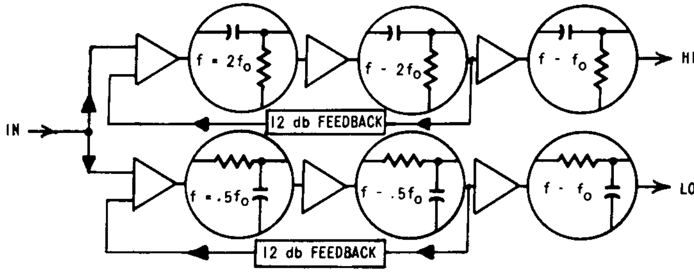
So now the distinction between package and component manufacturer has become somewhat hazy. Most manufacturers strive for the highest quality product at the price chosen by the customer. And they design with varying degrees of flexibility to suit both the customer who doesn't care how it works, as well as the man who wants to fiddle around on his own.

MONO VS STEREO

A few years ago, such an expression in a book like this would have presented a pro and con argument between monophonic and stereophonic high fidelity. The purpose here is to point up differences, because what makes a good mono system is not necessarily the same as what makes a good stereo system.

Before stereo became popular in the late '50s, high fidelity systems had been the subject of a power race. Many comments were made in both high fidelity and general interest publications to the effect that "fidelity" was being equated with "volume" or loudness. The enthusiast of those times would often comment that unless he played his system at such a high level, he could not hear the "finer nuances" of the program. An impression of dynamic range could be achieved only by turning the volume very high. In speaking of volume, we should differentiate between actual sound level as might be measured by a sound level meter, and the impression of volume conveyed to the listener.

Three factors convey an impression of volume to the listener: (1) the actual sound level; (2) the introduction of audible distortion; and (3) the use of more than one source for the total sound heard. The last item was responsible for the



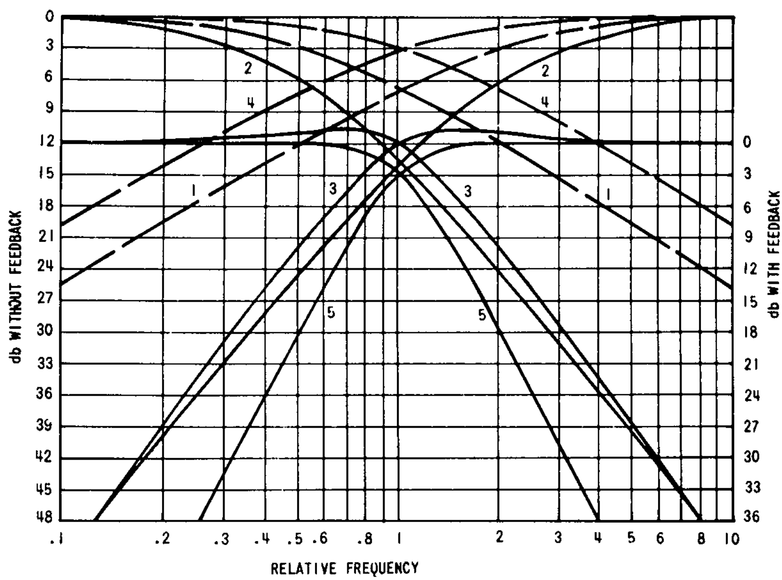
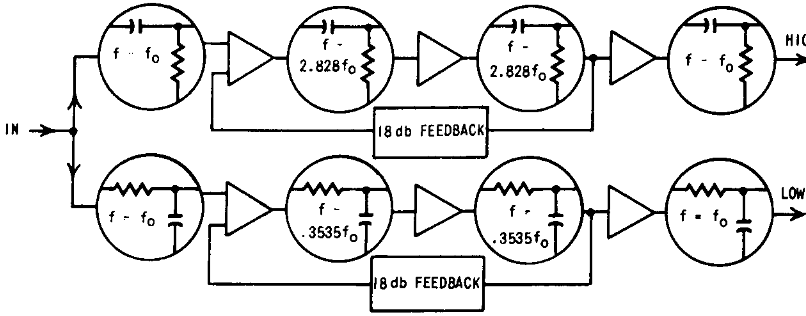


Fig. 6-9. Similarly, this method may be used to obtain a correctly designed electronic crossover with an 18 db/octave ultimate slope and a 9 db/octave slope at the crossover frequency. Top, a block schematic showing how the circuit is arranged; bottom, how the response is built up. Curves 1, the response of each roll-off within the feedback loop; curves 2, the combined response within the feedback loop before feedback is added; curves 3, the combined response after feedback is added; curves 4, the response of the additional roll-off outside the feedback loop; curves 5, the final response formed from input to the respective outputs.



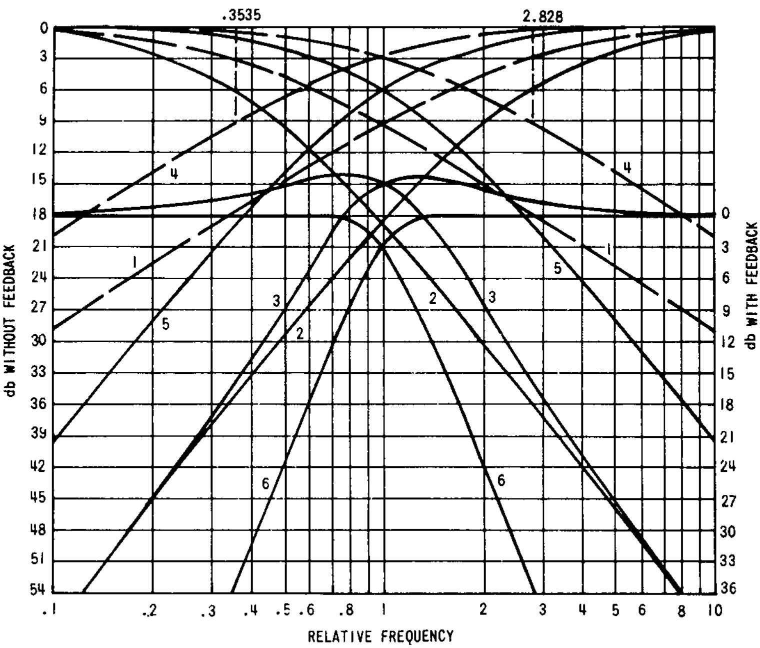


Fig. 6-10. These sketches show how to obtain a correctly designed electronic crossover with 24 db/octave ultimate slope and a 12 db/octave slope at the crossover frequency. Top, the block schematic; bottom, how the response is built up. Curves 1, the response of each roll-off within the feedback loop; curves 2, the combined response within the feedback loop; curves 3, the response external to the feedback loop before adding the external roll-offs; curves 4, the response of each external roll-off; curves 5, the combined response to the two external roll-offs; curves 6, the final response from input to the respective outputs. Notice that the triangles in the upper parts of Fig. 6-5, and Figs. 6-8 through 6-10, represent isolation between roll-off effects, with or without amplification.

ever more complicated speaker systems that developed before the advent of popular stereo, and for the trend to connect a number of speaker systems together and locate them around the room. This did enhance the impression of volume to a greater extent than it raised the sound level, and sometimes gave a better impression of realism, too.

The introduction of audible distortion has been responsible for the trend in what may be described as "teenage music." Many of the records for this market have distortion deliberately added during recording, because it makes the program sound louder. Teenagers tend to operate their sound systems with the volume at maximum. If a system is designed so that only little distortion can occur when the volume is at maximum, many teenagers complain that it will not "go loud enough." If some preset attenuation can be eliminated so the volume will allow considerably more distortion, they are satisfied. Asked to judge between a 5-watt amplifier operating with large amounts of distortion, and a 50-watt amplifier giving perhaps five times the actual acoustic output, the teenager is convinced the 5-watt system is the "louder." (And from the effect on our ears, we won't argue the point!)

But the enthusiast of those days was aware of distortion as well as volume. So he wanted it loud <u>and</u> undistorted. A multiplicity of speakers helped sometimes. But the only other way to get it loud and undistorted was to use a big amplifier and turn it up high. Few (except those who studied electroacoustics) realized that the difference between 5 watts and 50 watts is only 10 db in level—about 1/12th of the total range from the threshold of hearing to the threshold of feeling. So the effect of substituting a 50-watt system for a 5-watt system was to make it sound only slightly louder in the immediate vicinity, but to make it audible for more than three times the distance in any direction, including through walls! Needless to say, this was the era of neighbor complaints.

Stereo has changed all that. While feeding many sources into a single program sampling (channel) was some improvement, the use of two channels to provide stereo does far more toward creating a satisfactory illusion of loudness and dynamic range. At last enthusiasts are learning to be content with a sound level that will satisfy them without annoying the neighbors!

But this change has had some problems of adjustment subjectively. Habits die hard with some people, and some enthusiasts have actually induced deafness by the use of excessive level on mono. With stereo, not only is such high level unnecessary, but it actually can detract from the stereo illusion in most environments. Also, it is unnecessary to advance the bass and treble controls to maximum, as many did on monophonic, especially when they felt they couldn't play it loud enough for their satisfaction. Stereo gives a better illusion when the frequency response used is nearer to "flat." We shall discuss more about adapting the system to its acoustical environment in Chapter 9.

Chapter 7

Commercial Sound

The "commercial sound" category covers a variety of systems, including public address and sound reinforcement or relay, intercom, and paging. Each has somewhat different requirements to which the system must be tailored. And they all differ from the concepts employed in studio systems (covered in the next Chapter) in that they serve a purely service function, and thus are designed on a strictly economic basis.

PUBLIC ADDRESS

Sound reinforcement or relay systems are designed, generally, to augment natural facilities. In the old-fashioned auditorium a speaker stood on a podium and exerted his vocal cords a little more than most modern speakers do, and the building was usually built to help carry the sound of his voice to the audience.

As buildings became larger, and architects sought to draw attention to their designs, rather than to provide the functional quality of enabling the audience to hear what they came for, electronic aids have become more essential. Also, there always has been the person whose voice seems to diminish in inverse proportion to the number of persons being addressed — the soft-voiced speaker. Some of these people cannot even be heard in the front row of the auditorium without a microphone. So the job of the public address system may differ, not only from building to building, but in the individual using the microphone.

So, in one public address installation the problem may be to reduce reverberation, a condition that makes people with louder, faster-talking voices unintelligible, though very audible. In another it may be to enable the softer-voiced person to be heard at all. And sometimes both needs must be met.

"Relay" extends the audience area into another room, where the sound would not reach at all without artificial means. Where this is the object, reverberation or feedback is no problem because sound from the loudspeakers has no way of getting back to the microphone.

There are two more factors that vary the requirements of the system: one is its location, indoors or outdoors; the other is the kind of program to be handled, whether merely speech, from people with a variety of voices, or including music, which makes the system requirement a little more rigorous. Different auditoria vary tremendously in acoustic qualities, which must be considered in developing or designing a suitable installation. But the biggest difference of all is between an indoor and an outdoor installation. Not only does this affect acoustics but the quality of the equipment to withstand elements—the weather—is involved.

Indoor equipment is designed to be durable and of good quality, without the need to withstand the abuse of inclement weather, but outdoor equipment must not deteriorate as a result of such conditions. In designing equipment to withstand such weather it is inevitably more difficult to achieve the quality normally sought for indoor use. Fortunately, the better acoustic environment usually encountered outdoors permits this deterioration in quality to be acceptable under those circumstances.

MICROPHONES

The most important item in a public address system is the microphone, or microphones. In some respects, the quality of the microphone(s) selected is more important for this application than for studio use. A microphone should have a smooth frequency response, because acoustic feedback will emphasize any irregularities in response, even before the gain is high enough to cause acoustic howl (See Chapter 4).

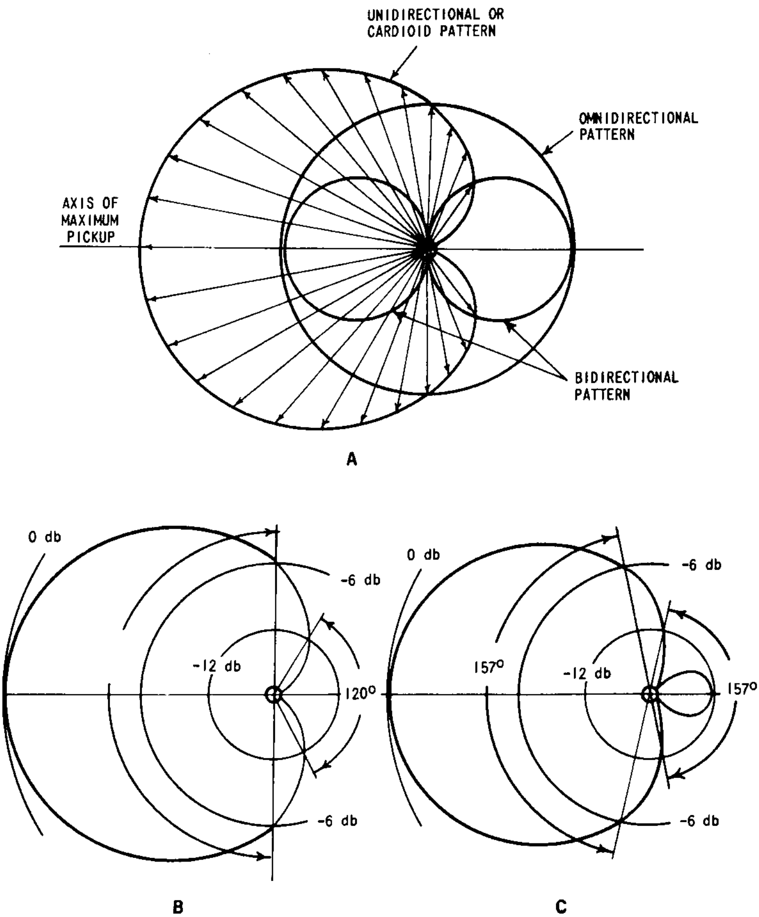


Fig. 7-1. These polar patterns show a variety of microphone directivity patterns. (A) the two basic patterns, omnidirectional (or nondirectional) and bidirectional that, put together in equal parts, build the true cardioid or unidirectional pattern; (B) various angles of attenuation in the ideal cardioid: a loss of 6 db occurs over an angle of 180° around the front and a loss of 12 db or more occurs over an angle of 120° around the back; (C) the improvement achieved by the supercardioid: the angle within 6 db is reduced from 180° to 157° at the front; the angle below 12 db is increased from 120° to 157° at the back.

Also, the irregularities will make the system howl sooner—that is, before as much active gain is available—so that the apparent sensitivity of the microphone is reduced when it has an irregular frequency response.

Directionality can help in public address situations, but it must be used correctly. The important thing to realize is that directionality is purely relative. A unidirectional mike picks up sounds from almost all directions—it just picks them up from the designated direction with a little more sensitivity than all the other directions. So don't expect directionality to work miracles!

For one speaker the unidirectional (cardioid) type is usually the best. However, where conditions are extra difficult an ultra-cardioid may be better. This sacrifices some of the rejection at the rear to achieve a narrower pickup zone in front, along with a generally lower sensitivity throughout a wider zone at the rear (Fig. 7-1). A directional microphone enables the person speaking to be at a greater, more comfortable distance from the microphone, and yet be adequately heard. Having to speak at a distance of 8 or 9 inches from a microphone can be very tiring, because it means the speaker must hold his head virtually motionless while talking, or move it in such a way that his distance from the microphone does not change.

For interview purposes, a variety of approaches may be adopted. The most common is to try and get both people in front of a single unidirectional microphone; this is not as easy as it seems. They cannot look at each other comfortably, and they must be quite close to each other for both of them to be sensibly in front of the same mike (Fig. 7-2). Of course, a regular cardioid is better than the ultra-cardioid for this purpose.

Another way is to separate the interviewer and interviewee by a considerable distance (Fig. 7-3), but this is not a comfortable arrangement either. If the mikes are brought too close together, both will pick up each speaker, and a loss of quality results.

Another arrangement not often used is actually better, but the participants need briefing, since it is different from other microphone uses. This employs a bidirectional microphone between the two people (Fig. 7-4). They can be at a comfortable distance from each other, can partially face one another

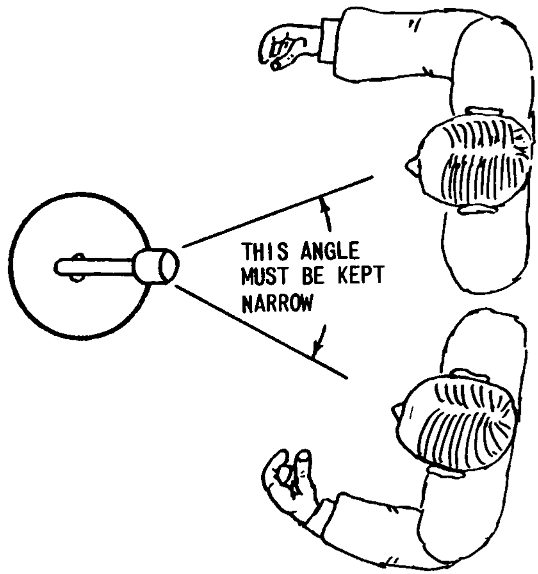


Fig. 7-2. This sketch emphasizes the problem of using a single unidirectional (cardioid) microphone for interviewing (see text).

and the audience, and both are at acceptable proximity to the microphone. But they need careful briefing to the fact that each of them is actually using a different sensitive point in the microphone's pickup pattern. Unless they understand this, one of the participants is almost certain to wander into the "dead" zone.

For a larger number of participants a variety of approaches

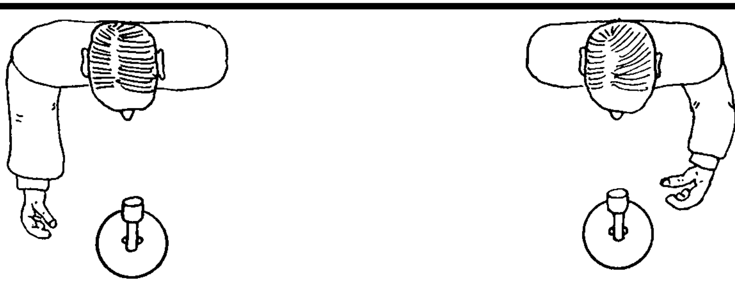


Fig. 7-3. An alternative interview method is to use two unidirectional microphones at a distance.

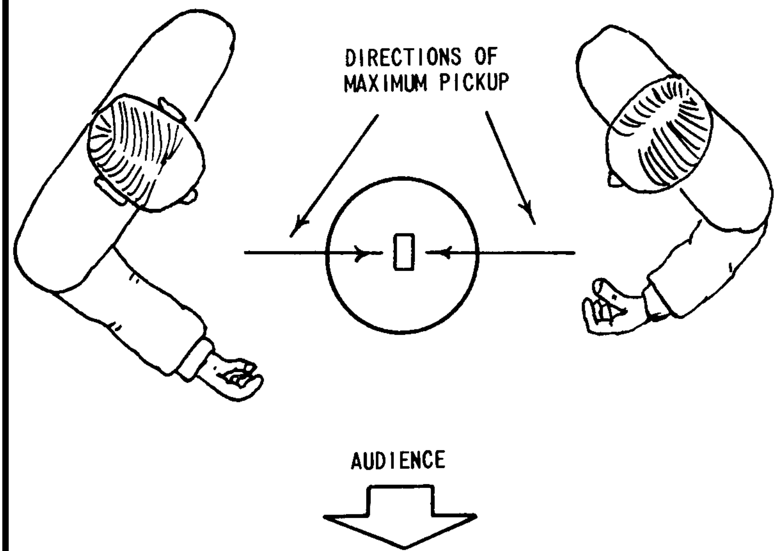


Fig. 7-4. Here, a bidirectional microphone serves a 2-participant interview pickup.

is again possible. One is the use of a single microphone of really high quality, used with high gain, at a considerable distance from all of them. Another is the use of individual microphones for individual (or groups of) participants. And a variant of this calls for bidirectional microphones used in a similar way, but this requires only about half as many microphones, and they should be spaced apart so that none of the participants is likely to be picked up by more than one microphone at the same time (Fig. 7-5).

When multiple microphones are used in a public address situation, beware of thinking that more microphones provide more complete coverage, because there are serious limitations to this concept. Unless the microphones are closely matched in sensitivity and impedance, they should not be connected to an input so that they are all controlled by a single gain control, since this may result in a more sensitive microphone causing acoustic howl problems before the others have even a useful pickup range.

And even when individual gain controls for each microphone (such as with a mixer) allow all of them to be set to optimum gain, that gain is seriously reduced by having them all "live" at the same time. Each will pick up a part of the standing-wave pattern that ultimately will cause an acoustic howl, so that each microphone has less usable sensitivity than when they are used individually. For this reason, a complicated program, such as a play, using multiple microphones should be controlled by an operator provided with cues that enable him to have each microphone "live" only when it is actually needed. This serves as an additional safeguard against picking up unwanted "casual" remarks that may be made off-stage.

Another situation is the conference or convention where participation from the floor is expected. To provide for this, microphones are placed within reaching distance of delegates at strategic points on the floor. Here again, it is highly desirable that these mikes should be "live" only when actually in use. Being out on the floor makes them even more liable to cause acoustic feedback than platform or stage mikes. And being in the audience area makes them susceptible to accidental pickup of sounds not meant for the general audience, such as a remark by people finding their way to or from a seat.

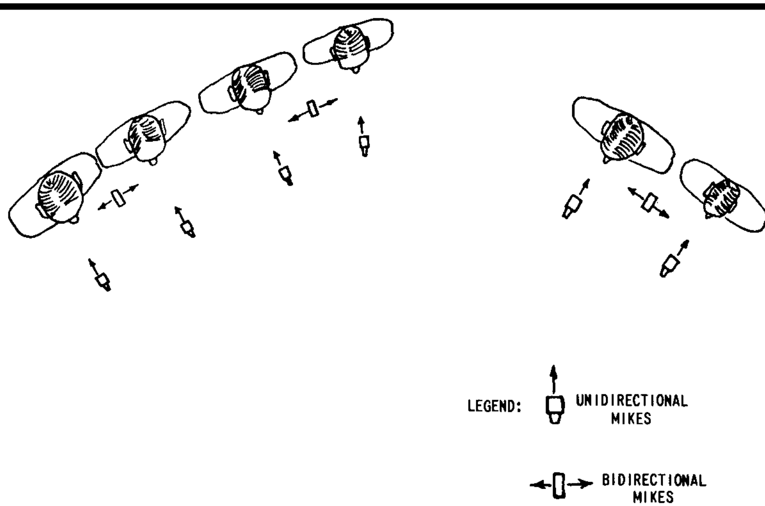


Fig. 7-5. Here are two ways of covering a larger group of interview participants: with single unidirectional microphones, or with half as many bidirectional microphones. Of course, both techniques are not used simultaneously but are alternative methods of covering the same situation.

Another way to handle this situation, which involves few additional operating problems and a great deal more flexibility, is to use a "machine-gun" mike—one of the newer super-directional types capable of aiming a great distance within a narrow angle. Such a mike, operated from a spotlight window (Fig. 7-6), can cover the entire audience area, one person at a time. It is not even necessary for the participants to leave their seats.

Wherever a number of microphones are used, so that they need to be rendered active only when necessary, it is essential to provide an operator-control position where all the mike positions can be clearly seen. For the conference situation it may be worthwhile providing the chairman with mike controls, which should be of the simple on-off type, preferably with spring-return keys. This way he can select the appropriate microphone to be used by participants he recognizes on the floor. This is much better than having the microphones switched locally. With the latter method someone is always either leaving a microphone "on" after he has finished using it, so that when another mike is used acoustic feedback may be encountered, or else a participant does not realize the mike he is speaking into is "off" and he starts to speak without realizing he is not being heard. Having the mikes under the control of the chairman, or an operator who takes his cue from the chairman, is the much better way. (See Chapter 4.)

Microphones for use outdoors do not have the feedback problem because there are far fewer reflecting surfaces. For these applications the more important feature of a microphone is its ability to reject wind noises. The only sure way to check this is by comparative test, using the windshield provided with the microphone. The function of a windshield is to divert the sound around the microphone in such a way as not to create audible sound products, such as whistling, while permitting the desired sound waves picked up to reach the microphone. The effectiveness of such a device can be checked only under operating conditions.

For music, indoors or outdoors, the microphone needs a wider and smoother frequency response than for speech. For speech only, the low-frequency end of the response can be curtailed, often to advantage, because it stops rumble-type effects. Where a system is to be used for both, provision of

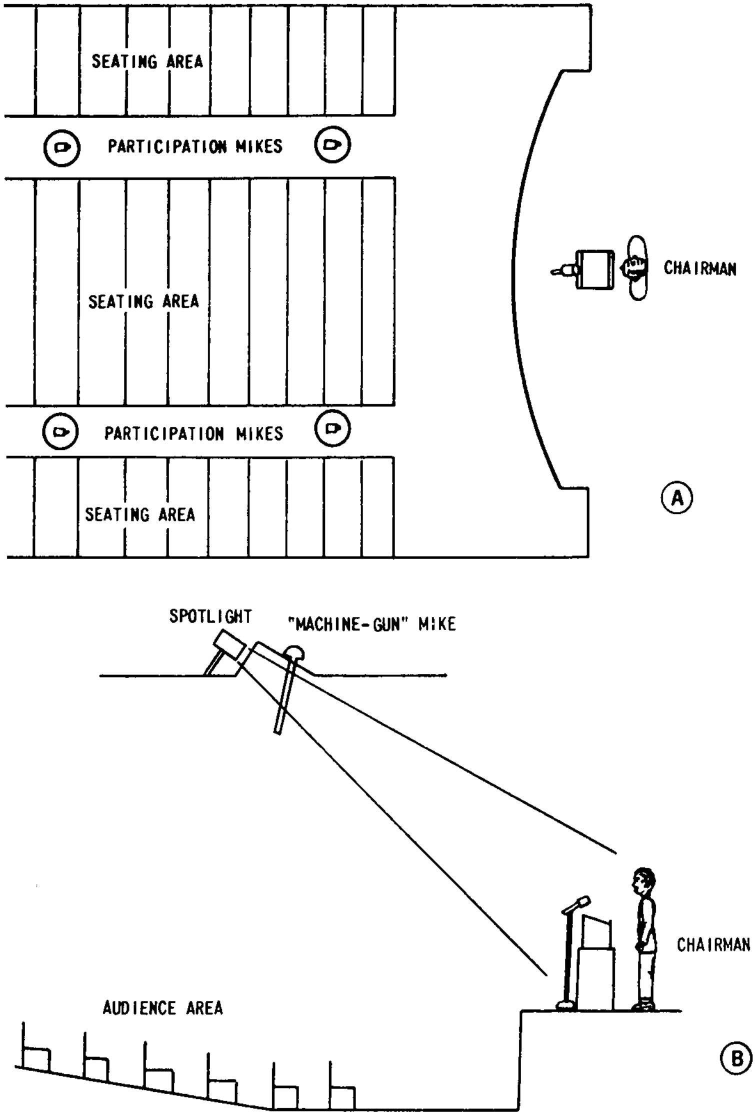


Fig. 7-6. Alternative methods of providing for audience participation: Microphones are within a reasonable distance from different parts of the audience seating area in layout A. At B a "machine-gun" mike, located in the spotlight balcony can be aimed at any individual member of the audience while in his seat.

a music/speech switch that introduces a cut-off below about 250 Hz, may be an advantage. Voices will sound clearer in the speech position, but music will be bass-deficient. The music position will restore the bass.

In difficult installations the provision of individual tone controls can help make the best of each person's voice (Fig. 7-7). To aid in correcting for a variety of voices participating in a discussion or dialog in a play, a two-position switch to select one of two sets of tone controls (or more, if desired) is a worthwhile refinement. Then the operator can switch this control each time the other person speaks, so the tone control is optimized for each voice. Some voices sound high-pitched without such correction, others are boomy. Under normal microphone conditions these differences would not show up to such a marked extent. But a public address system tends to emphasize these differences, as well as the irregularities in mike response, building resonances, etc.

LOUDSPEAKERS SYSTEMS

Coupled with microphones in public address work is the choice of suitable loudspeaker systems. This applies not only to a wise choice of speakers, but also to how they are located in the audience area. Where reverberation is a problem, more speakers are needed, placed so that anyone in the audience is within a fairly small distance from the nearest speaker.

Then, the speakers are operated at low levels, called low-

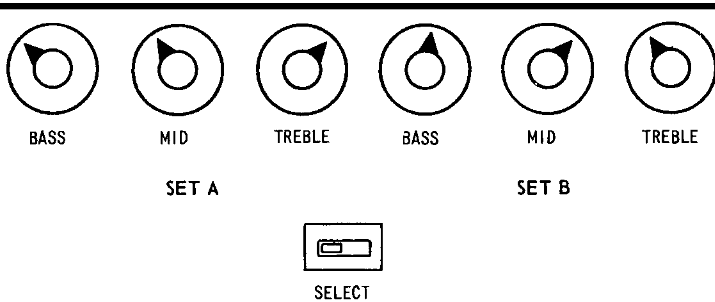


Fig. 7-7. Simplified layout of multiple tone controls, designed so that alternate settings can be repetitively used in sequence.

level distribution, which offers two advantages: 1. each member of the audience hears only the speaker nearest to his listening position, directly; 2. the level of the reverberant energy is reduced considerably, and sound is more intense in actual audience areas than in the empty part of the building. Where reverberation is not a problem, where good absorption prevents the buildup of echo effects, fewer speakers are necessary with greater economy in installation costs.

For outdoor installations, speaker locations are often dictated by natural service area boundaries. Speakers may serve whole stand sections, in a stadium for example, provided this will not result in some people being deafened, while others have difficulty hearing because of the disparity in distance from speakers.

For indoor installations, cabinet speakers, or horns that at least use plastic trumpets, should be used, because metal horns produce a trumpety sound that any indoor installation will emphasize in the acoustic feedback effects. For outdoor installations, metal horns are ideal, because they have a durability and resistance to the weather that indoor types don't have.

PA AMPLIFIERS

Amplifiers for public address systems need to be "workhorse" type units. Rugged, reliable, and of good quality, but not necessarily the performance quality expected of the better high fidelity systems. In fact, some limitation of frequency response is probably desirable, because the extremes—very low and very high frequencies—add little to the realism or fidelity, but can add to accentuation of undesirable effects, such as rumble and hiss.

PA speaker efficiency has a direct bearing on amplifier power requirements. The difference between using speakers averaging 2% efficiency and 20% efficiency amounts to a 10:1 change in power requirement. Where a 50-watt amplifier might adequately serve quite a large area, using the more efficient speakers, 500 watts worth of amplification would be needed to produce the same amount of sound from the less efficient speakers.

An additional factor in determining power requirement is the background noise with which the program must compete. Estimating this can be very deceiving because of the logarithmic sensitivity of human hearing. In the absence of certain high-level background noise a system may give the impression of being more than adequate, but when the high-level background appears the system seems suddenly very inadequate. Suppose in the absence of the background noise the system produces a level of 60 phons. This sounds quite loud against a quiet background. Probably turning it down to 50 phons—one tenth the power—will make listening more comfortable. But now assume a background at 90 to 100 phons appears. The 60-phon level becomes virtually inaudible. The level at the listening position must be raised to 70 or 80 phons to achieve audibility, and even this will not sound loud while the background level is present.

Background level comes in a variety of forms. The audience themselves may provide it, for example at a race track when each is shouting encouragement to his favorite horse. Or it may come from some external source, such as a nearby industry, commuter track, or subway. Low-flying aircraft approaching or leaving a nearby airport may be another source of noise—one that may vary according to wind direction, and which runways are in use.

If you are sitting in the stands of a stadium when an aircraft passes, or a subway train comes by, you will notice, unless the operator changes level to offset the effect, that the sound system seems to get quieter as the interfering noise gets louder. It doesn't really change its power level, this is a subjective effect, due to masking. But a system that encounters this problem needs the power reserve to take care of it. Don't forget that, as well as extra power from the amplifiers, the speakers need to deliver the extra sound level to the audience.

A 100-watt audio system may feed a stadium with more than adequate power while the background noise is absent; in fact, it may operate at a 10-watt level, or less. But when the background noise presents itself, the level may need to be 1,000 watts or more, perhaps even 10,000 watts would be barely adequate. And probably the speakers could handle no more than 200 to 500 watts. This is when the level at the listening positions becomes important. The method of distributing sound that was adequate in the absence of background noise is no longer sufficient. You may need to employ the same technique used in low-level distribution, although it would be a misnomer to call it that for this application: Move the speakers in closer to individual segments of audience, and use many more of them. This gets the level higher where you need it.

BACKGROUND MUSIC, INTERCOM, AND PAGING SYSTEMS

Such systems also come in the general category of commercial sound, although they are quite a different kind. In some instances they serve a function very similar to public address, if in a different environment. Such systems are finding use in an increasing number of establishments—offices, factories, stores, airports, and other public buildings.

While background music may at first seem to be merely an amenity, it has proven economic value. "Music-while-you-work" was introduced during World War II because it was found to boost production at a time when there was a manpower shortage. In stores, background music puts buyers into a happier mood, and the sales go up. In airports and other public buildings, it may have no direct economic advantage, but as background music is now readily available, and a paging system is virtually a necessity, the surroundings can be rendered more pleasant by utilizing the system for background music, while it is on standby.

Background music is chosen to be unobtrusive, not requiring listener concentration for its enjoyment. And the reproduction is not so wide-range as either high fidelity in the home, or musical program broadcast or relayed exclusively for entertainment. Reducing the response to lower frequencies enables smaller speakers (physical size) to be used, such as are suitable for the voice broadcasts, and reducing the upper frequencies lends a pleasant "roundness" to the reproduction that helps ensure its unobtrusiveness.

Paging systems fit in quite naturally with background music, using microphones at certain control locations for feeding necessary paging announcements into the system. At an airport, for example, each airline will have a microphone to enable the counter to page customers who may have failed to pick up their tickets or other material, or for whom telephone messages may be waiting. Such announcements are normally carried over the entire airport. However, in a large airport, boarding announcements are not required throughout the entire complex, only in the concourse or part of the airport operated for embarkation of the particular airline. So the announcement, "North-West Orient Airlines, Flight 25, now boarding at Gate C-14," will go out over concourse C, which includes the area where North-West's passengers are likely to be lounging around.

This requires a quite complex system, but it is built up of relatively simple units. Microphone inputs feed into immediate microphone amplifiers, and power amplifiers feed sections of loudspeakers, and the outputs from groups of microphone amplifiers are fed to the appropriate power amplifiers so the correct groups of areas are covered with each group of announcements. One more thing in such a complex system: the indication whether the system, or any part of it, is free for a particular announcement. This is achieved by a system of lights, keyed by any microphone in use, to indicate to all other microphone stations that might want to use the same group or groups of speakers that the system is in use. A more sophisticated system might include an electrical (by means of relays) or electronic (by means of signal-gating circuits) interlock to ensure that accidental pressing of another key will not interfere with an announcement in progress.

Intercoms have one fundamental difference from the other systems, although they may sometimes be combined with them: the ability for two-way communication. An intercom, except for the simplest from-the-kitchen-to-the-front-door type, has the most complex circuitry of any audio system. So much is this so that it has become a specialized field in itself, similar to the telephone service. For this reason we will not go into extensive details here, beyond stating the variations between systems.

An intercom system is made up of any number of stations, which fall into two categories—master and slave. A master station can initiate a call to any other station. A slave station can initiate only a call to its own master, or a limited number of masters. A master station usually has the capacity to address all the slave stations at once, when the system behaves in exactly the same manner as a paging system. This can be achieved by pressing all the slave-station call keys at once. But usually an extra key provides for making such a blanket call without the need for pressing all of them.

The simpler intercom systems allow speaking only one way at a time. Each is equipped with a press-to-talk switch, which is usually the only switch on a slave station, but is additional to the station selector switches on a master station. Using the one-way-at-a-time approach not only saves on amplifier requirements, but it enables the same transducer to serve as both loudspeaker and microphone. This is usually a small unit designed for the purpose and having an impedance of 45 ohms.

Such an intercom does not allow conversation with face-to-face freedom because the other person cannot speak until the initiating speaker releases his press-to-talk key. So if the party on the other end launches into a lengthy explanation that doesn't answer the intended question, you can't interrupt with, "Hold it, I wasn't asking that!" You have to wait until he's through, and then tell him, in effect, that all his explanation was a waste of time, because he misunderstood your question!

In these modern days, when time is at a premium, a more complete two-way system is desirable, one that allows both ends to speak at once if need be. This means that each station needs a speaker and a microphone, as well as amplification for each. Also, closing both circuits at the same time introduces the acoustic feedback problem which was absent from the one-way-at-time system. In most systems of this type the problem is overcome by using the signal level to control gain. When the microphone picks up speech the signal is used to boost the gain of the microphone amplifier and to correspondingly cut the gain of the speaker amplifier. Thus, the overall gain doesn't change, but the gain is always higher in the direction at which speech is for the moment traveling. If the other party wants to interrupt, it is natural, even in conversation, to raise one's voice a little. This equalizes the gain, so the interruption is heard. Correctly adjusted, such a system provides a quite natural conversation link between the points it serves.

Many intercoms also can be used for extempore conferences, without the need for the participants all to come into one room.

This is achieved by one of the masters depressing several station selectors at one time. Some intercoms have few, or even no slaves. Every station is a master, capable of calling any other station on the system. The wiring for such a system is a little more complex, but the circuitry is basically the same.

Any system with a large number of masters is accompanied by a privacy problem. If more than one station selects one of another group of stations already in communication, unwitting listening-in can get started. The solution to this is privacy keying, where both stations select only the other on their station selector. This then automatically closes both of them to interruption from any other circuit. Having gone this far, on the development of a system that now has no "slaves," the equality of "status" can become a nuisance. It may be impossible for the boss to get a message through, because the storeroom boy is taking an undue amount of time trying to date an office girl. So in this kind of system the master station acquires a distinction from the rest, different from that defined earlier. This is simply an over-riding circuit that allows master stations to indicate that they are calling, by means of a buzzer and light, or some other inescapable signal, enabling the conversation of lesser importance to be terminated as quickly as possible, so the more important message can get through.

With such a variety of intercom systems available, obviously each has its own area of usefulness, and the system's engineer will need to determine just which system best suits the needs of each client.

Chapter 8

Studios

The philosophy behind the audio systems covered in the previous Chapter and those discussed here is governed by rather different requirements, not the least of which is economic. The reason the perspective changes so dramatically with studio equipment stems from the part played in the overall picture. In commercial audio systems the number of people served varies from a few hundred to a few thousand. But records made on studio equipment, or the audience served otherwise, may run into millions. For example, while not every record sells millions of copies, every company is striving for those that will for obvious reasons. A record that sells only a few thousand copies may not reach the break-even point. So any device that will help make bestsellers is worth its weight in gold, almost literally. So the cost of such devices, which would seem disproportionate for any other application, may soon pay for itself in studio equipment.

MICROPHONES

Since nearly everything handled by a studio audio system has to "come in" through one or more microphones, microphone quality is of paramount importance, although for a somewhat different reason than for commercial installations. In a public address system the microphone needs various quality features to provide optimum performance against problem conditions. In a studio system the microphones need qualities that give the ultimate in naturalness of reproduction—the maximum fidelity. Thus, generally speaking, directivity is an asset for commercial sound use. And the directivity should be good at all frequencies. While the frequency response should be smooth, this requirement should be second to directivity.

But in the studio, directivity is far less important. Directivity may be needed for microphones in television or motion picture studios, where the microphone must be kept out of the way of cameras. But for simple recording, directivity is not important because the microphones can be placed where they need to be, and feedback is no problem. It is because microphones are so important to the end result that studio managers and operators are very critical about them. Of the available good microphones, most people in the business have formed a preference that is difficult to shake. In most instances the preference is good, but sometimes it's based on earlier experience, and aural conditioning.

"COMMERCIAL" SOUND

Now that fidelity has reached quite a high standard, most studio people prefer microphones that give the most "transparent" effect: make a recording where the illusion is so good that the original performers seem to "come through" completely. But some of them talk of "commercial sound" —which has a different meaning from the systems discussed in the previous Chapter—as they personally recognize it. Such sound is "commercial" to the extent that a segment of the public has a preference for the same audio qualities, consciously or subconsciously. With some it's just the right amount of bass resonance to give a certain kind of boominess. With others it's just the right amount of peaking before high-frequency roll-off to give a certain "crispness" or edginess to the reproduction.

These notions of "commercial quality" are a hangover from the days when reproduction was essentially different from the original performance, and never likely to be mistaken for it. It's very like some of the radio receivers that might be advertised as having a "cathedral tone," meaning the speaker had a boxy resonance! But a long conditioning to this kind of sound as "good reproduction" can actually lead to an attitude that interprets more realism as "dull." The better realism, as it has become possible, has always needed "selling" because of this conditioning element. Greater realism often gives the initial impression to many people that something is lacking. If reproduced music should have a "boom boom" or a "tsh tsh," or both, then it seems lacking if you can't hear those effects.

But preferences change gradually, and usually in favor of the more realistic effect, because that, ultimately, is the easiest to listen to. Artificial effects ultimately become tiring. What may have started as a wonderful sound does not continue to satisfy because it cannot be identified as something real. This has been one of the problems that electronically-generated forms of music have had in gaining acceptance. And studios are now getting more and more involved in these devices. We are living in days when purely electronic sounds, that are not intended to simulate some older sound form, are beginning to gain acceptance in their own right.

STUDIO ACOUSTICS

Closely associated with the characteristics of microphones is their use to achieve the desired overall effect, acoustically. For any form of program that comes from live performers there is a natural acoustic environment which covers a range of permissible reverberation time. For example, a choral recording benefits from the addition of a fairly long reverberation effect, simulating a large cathedral echo. Smaller, intimate groups need a shorter time, again to simulate the environment in which the listener will imagine them when he hears the reproduction.

Environment can have other qualities, beside simple reverberation, or echo time. Different rooms or studios vary in the tendency to emphasize certain frequencies. For example, a room with plaster walls and ceiling and little absorbent material, even though it is too small to possess appreciable reverberation time, has a brilliance because of the greater efficiency with which higher frequencies are reflected. On the other hand, if the walls are wood-paneled, this will add character that is warmer, less brilliant. And adding really absorptive material—drapes, acoustic tile, carpeting, absorptive furniture—can make the environment sound "dead" or "quiet." In a good recording, using a high quality microphone, these qualities "come through."

How much they come through depends on the way the microphone(s) are used. Putting a microphone close to each performer tends to exclude or reduce environmental effect, while performing at a greater distance from the microphone, as when a single mike is used for a group, tends to emphasize the environment unnaturally, especially in a single-channel reproduction, and stereo changes this a little.

Environment matters in two ways: to the performers and to the ultimate listeners. To the former it can affect actual performance. To the latter it's part of the overall product, and thus may affect sales. A musician is affected by the room in which he works. Often engineers don't appreciate this sufficiently. If you don't believe it's true, just try ordinary conversation in an echo chamber when you have an opportunity. It's uncanny, and you're glad to get outside where your voice sounds "normal" again. This is perhaps an extreme example, but it shows that the sounds we make are affected by our environment, and musicians also are subject to this.

Playing in an over-dead studio makes a musician less able to hear his own performance, and less able to relate his performance to that of the group. He becomes more mechanical, less sensitive. Playing in an over-live studio also has its disadvantages, of course. The musician can't separate his performance from the sound he, or they, made a few seconds ago! So every studio has a range of ideal environment characteristics, from the viewpoint of the performers, enabling the sensitive musician or other performer, to do his best.

For stereo, as opposed to the older single-channel (monophonic) productions, the ideal recording should have less reverberation included in each channel than was acceptable for good monophonic effect. But, of course, some is required. The best effect must be determined subjectively. For this reason, most studios prefer to err on the side of recording too little reverberation, either by using a studio that is on the "dead" side, or by using enough microphones close to individual performers to minimize pickup of what reverberation the studio has. In this way reverberation may be added

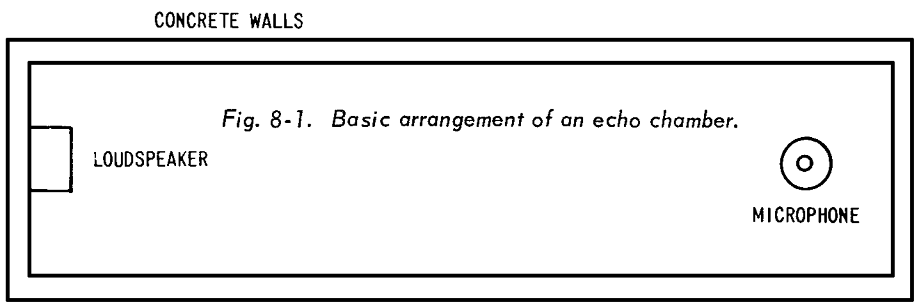


Fig. 8-1. Basic arrangement of an echo chamber.

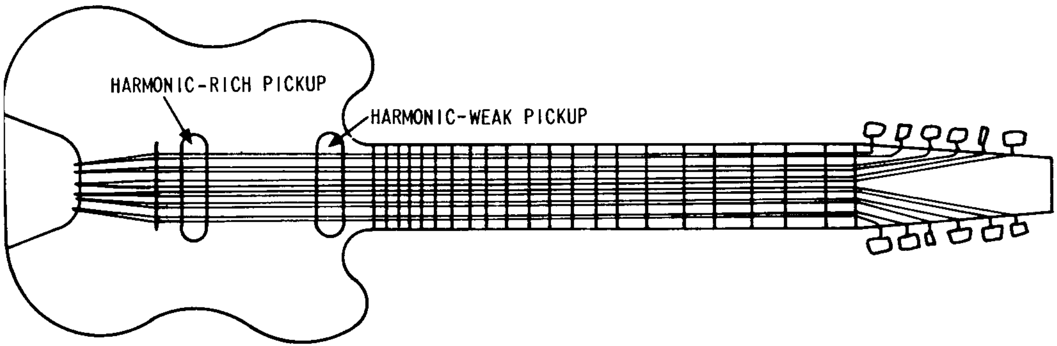


Fig. 8-2. Pickups on an electric guitar or bass are frequently arranged as shown here to provide outputs of different timbre, from which the ultimate sound effect can be given quite a varied range.

"to taste" after the original performance has been mastered. This extra reverberation may be made either with an echo chamber, where the original is reproduced by a loudspeaker at one end of an echo chamber, and the sound picked up again by a microphone at the other (Fig. 8-1), or with a tape reverberation simulator.

A simple tape recording, with playback heads at short intervals after the record, produces something resembling an echo effect, but it does not have the character of any real building. So echo chambers came to be preferred to the artificial variety. However, as electronics have become more sophisticated, and solid-state circuitry has allowed numerous variations in quite compact space, the artificial reverberation method has developed a flexibility not possible in the echo chamber. So the trend is now toward the artificial means of reverberation treatment. Also, multiple pickup heads feeding separate preamplifiers, each with tone controls, allow the introduction of unique coloration to that particular element of "reverberation." In this way a close synthesis of almost any desired natural reverberation can be made.

ELECTRONIC MUSICAL INSTRUMENTS

Musical instruments employing electronics can be divided into two categories: those whose tone generation is basically conventional, but which rely on electronics to amplify and modify the resulting sound; and those whose generation of sound is totally electronic, there is no conventional tone development at all.

Typical of the first group are electric guitars and basses. The tone is generated by the strings, but the instrument is not at all like its conventional counterpart which uses natural resonances in the design of the instrument to develop the characteristic sound generated by the vibrating strings. The electric instruments use a physical structure that is essentially resonance-less—rigid—so the strings produce little or no sound for natural acoustic propagation. Electrical pickups close to the strings, and disposed so that the harmonic structure generated can be varied by mixing the pickup output (Fig. 8-2), enable variable tone quality to be injected into the instrument's amplifier (some use stereo amplification). Then, additional effects are built into the amplifiers, such as electronic vibrato, reverberation, and tone control. Another addition is the "fuzz box" (also known by other names), the object of which is to add deliberate distortion to the sound. After all the effort that the older generation has put into reducing distortion to the lowest possible level, it is a little disconcerting to find the younger generation getting delight from putting large lumps of distortion back in!

In the studio, until recently anyway, electric instruments have performed exactly like any other, and miked in the same way, too. Thus, for overall stereo recording of a group that includes these instruments, close miking may be employed, with some form of artificial reverberation applied to the overall program "to taste." The studio's own reverberation could be eliminated altogether by using an electrical output from this type of instrument, instead of miking it. Two factors have discouraged this technique up till now: (1) the instrument's amplifier has not provided an output convenient for making a direct connection into the audio system; (2) eliminating reverberation from electrical instruments in this way, while not being able to do the same with conventional instruments in the same group, destroys apparent "balance."

As a bigger proportion of instruments become electrical, the second reason disappears. And when makers of electrical instruments provide outputs at a level suitable for connecting to a studio mixer system, this will be a much better way to achieve the desired result. In this way, the studio can "add" the acoustics that suit the musicians' needs best, and they

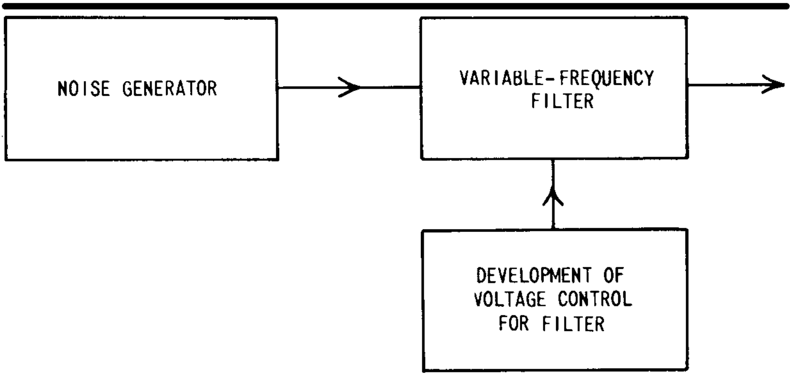


Fig. 8-3. Basic arrangement of the noise-type synthesizer generator.

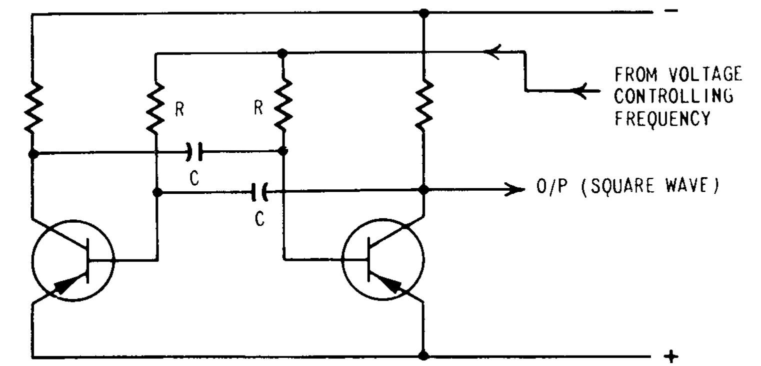


Fig. 8-4. Basic multivibrator circuit showing frequency-controlling elements R and C. Varying the voltage applied to the "top end" of resistors R changes the working frequency of this circuit.

will play by listening to the sound from the speakers built to go with their instrument amplifiers. Whatever kind of a "din" this makes for them to work in, the studio inputs can receive the individual contributions from each musician quite independently. Then the audio people can take this composite —or even the separate instruments, possibly recorded on multi-track tape with as many tracks as are needed—and apply reverberation and other effects, as desired.

ELECTRONIC MUSIC SYNTHESIZER

Another studio technique employs the all-electronic type of music synthesizer. The basic elements of a synthesizer are circuits that control frequency and amplitude by means of varying voltages. The basic generator may be either a source of "noise"—random movements of charge at low level, usually of the type known as thermal noise, amplified to a level recognizable as the background hiss that audio engineers have labored for decades to get rid of—or a tone generator of either sine or square waveform.

In the case of the noise generator, frequency is selected by passing the signal through a voltage-tunable filter that colors it according to the frequency selected (Fig. 8-3). In the case of a sine- or square-wave generator, frequency is selected by changing certain constants in the frequency-determining

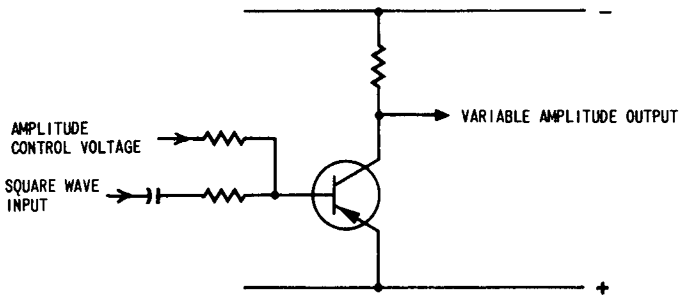


Fig. 8-5. This simple circuit can vary a square-wave generator output from zero to full voltage by applying a variable control voltage.

part of the oscillator. A sine-wave oscillator may use either a tuned circuit or a phase-shift oscillator circuit. And either one can be used as a frequency-selective amplifier by working it below the amount of feedback needed to cause oscillation.

The frequency of a tuned-circuit oscillator can be adjusted either by a variable-reactance circuit or by a solid-state reactor, such as a voltage-variable capacitance diode. In either case, a variation in reactance value is possible only from a ratio of 2:1 up to about 5:1 (reliably). As the frequency of a tuned circuit is proportional to the square root of the value of one of the tuning elements (L or C) the maximum change in frequency will never be appreciably more than an octave (2:1 frequency).

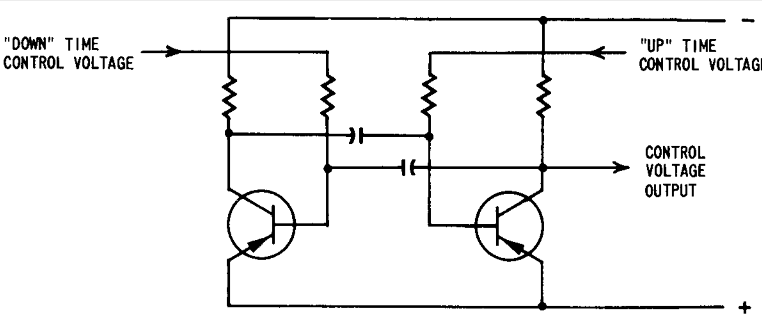


Fig. 8-6. Schematic of a flexible variable "up" and "down" time generator to provide the basic control voltages used for controlling amplitude and frequency.

If more than one element is used in a phase-shift oscillator, which may also employ variable resistors (varistors), the frequency change is directly proportional to the change in each element value. Thus, it is possible to vary frequency over a range of a little over two octaves in this way. Tuned circuits work on resonance between two reactive elements of opposite kind, and phase-shift oscillators and filters work on the phase shift of resistance/reactance combinations. Each depends on how the reactance of a component varies with frequency.

The square-wave oscillator uses an altogether different principle that does not directly involve reactance. Rather, it is based upon time constants. Applying variable voltage to change the operative time of elements whose basic time constant does not change can achieve a variation in trigger time, and thus frequency, of about 20 times, or over four octaves, which is comparable with the frequency range of the manual of an electronic organ.

Fig. 8-4 represents a simplified form of multivibrator (a more sophisticated form is needed to cover the 4-octave range); varying the voltage fed to the "top end" of the base resistors will change frequency at will. Amplitude may be controlled by one of two methods: with a sine wave the only way is by varying the gain of a stage. In tube amplifiers this was readily accomplished by means of a variable-mu tube. Up to 30 or 40 db change in gain could be achieved. In transistor circuits this kind of change is not quite so simple. And even where it is possible, a circuit that will go completely "off" is an advantage, which usually represents a gain change of more than 40 db.

The square waveform lends itself to a change in amplitude by variable clipping. A relatively simple circuit can then vary the output from zero to full amplitude (Fig. 8-5). Now, it's merely a matter of programming voltage-variable devices to control frequency and amplitude in any desired fashion; for example, additional multivibrators with different basic time constants, and variable-voltage time controls (Fig. 8-6) with the output modified so the voltage can happen either suddenly, or gradually, or with a pulse that returns to its original point

(Fig. 8-7). That much gives us the basic ingredients for making a square-wave variable-frequency, variable-amplitude generator. Several can be combined into a single synthesizer to produce a more complete musical output. But all the tones so produced will have similar quality. A square waveform has all the odd harmonics, in descending magnitude (but not very rapidly) which can be modified in sound to some extent by the application of formants that either accentuate or diminish the relative harmonic content.

The relative harmonic content can be changed more specifically by deliberately making the square wave asymmetrical in definite proportions. Once the wave is asymmetrical, even harmonics start to appear, and the family of harmonics generated changes in quite a complicated fashion, according to the nature of the asymmetry. Fig. 8-8 shows how the relative magnitude of successive harmonics varies as the ratio of "up" to "down" time changes, each expressed as a fraction of the fundamental component. Making the waveform extremely un-

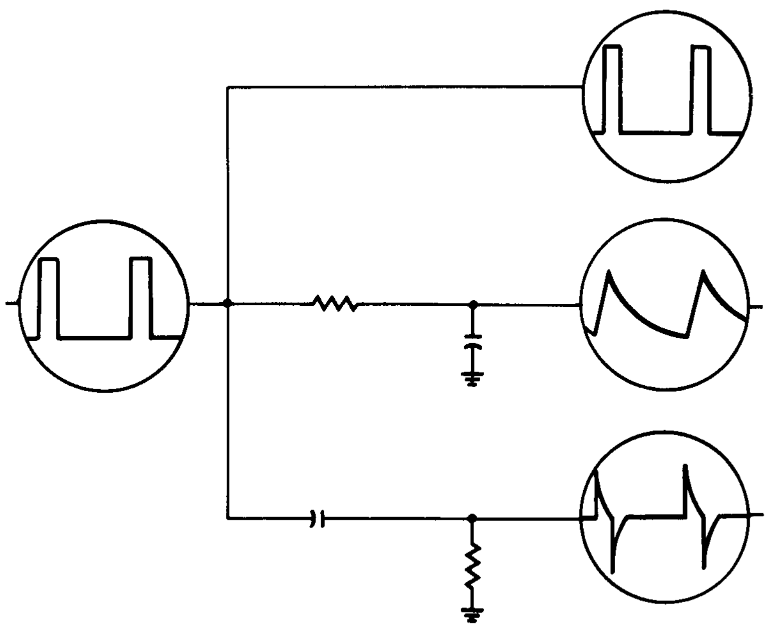


Fig. 8-7. These waveforms illustrate how the voltage developed can be modified to realize a variety of effects.

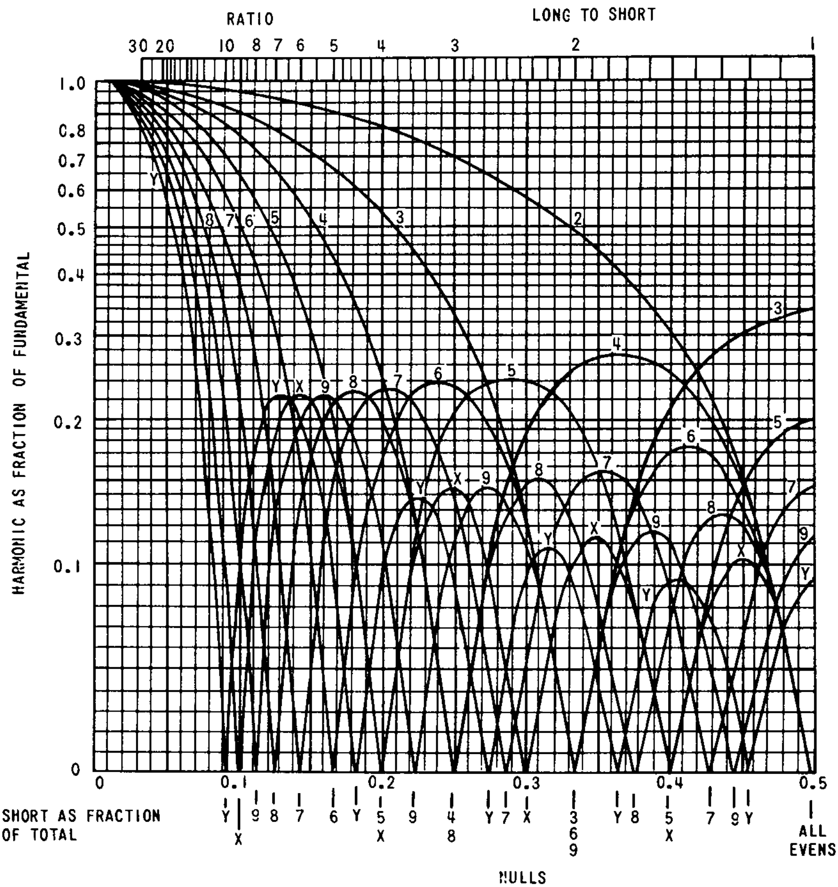


Fig. 8-8. Chart showing the variation in harmonic content obtainable from a square wave, where the "up" and "down" parts vary from the uniform (symmetrical).

balanced, which produces a very reedy effect, causes all the harmonics to have an amplitude almost equal to that of the fundamental.

Asymmetry in a square wave is easy to produce, either by using different component elements in the time-constants that determine the respective up and down trigger times, or by using separately adjustable voltages (Fig. 8-9). However, whether the shape of the square wave is changed, or whether the relative magnitude of its existing harmonic content is changed by formants, a whole family of tone colors is miss-

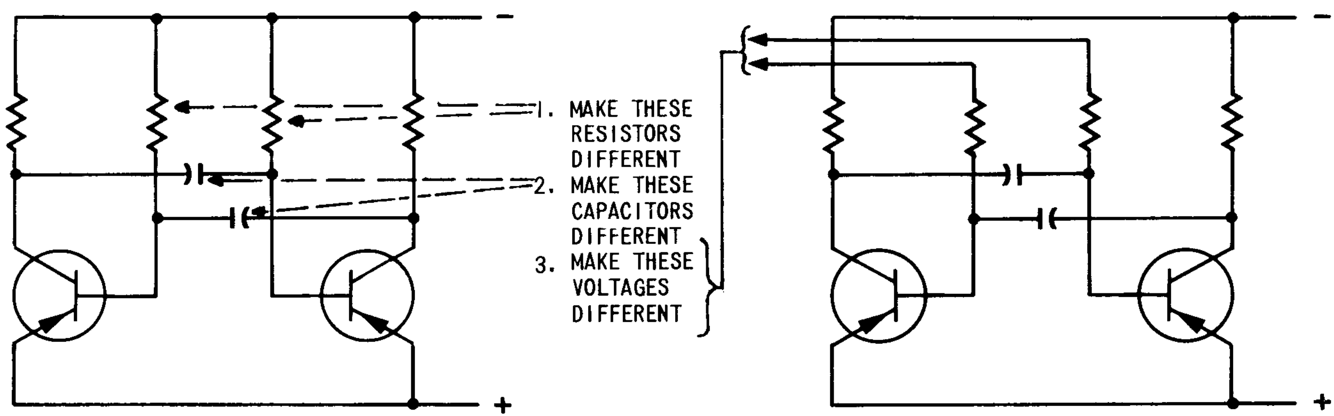


Fig. 8-9. Illustrated here are techniques of producing asymmetry in a square wave. The third way allows the proportions of the wave to be voltage-controlled, like its amplitude and frequency.

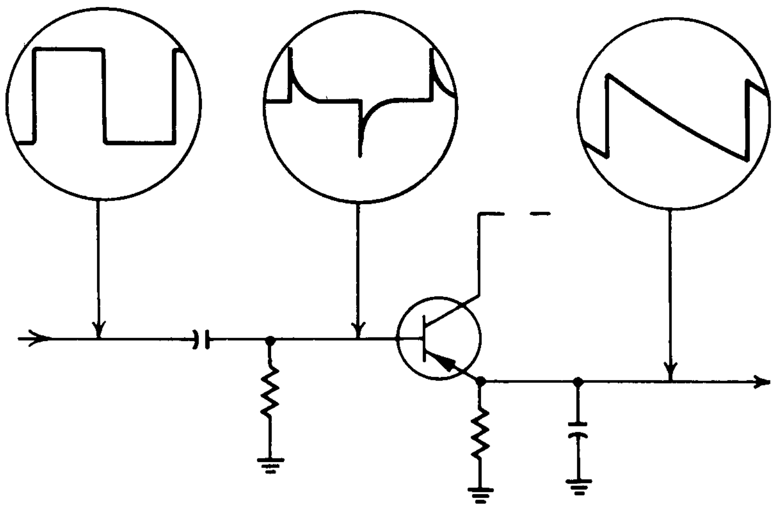


Fig. 8-10. This circuit that will convert a square wave to a sawtooth.

ing—that which starts with basically all the harmonics, instead of merely the odd ones, or a rather complicated variation, as the symmetrical and asymmetrical square waves do.

One way to correct this, or to derive the remaining group, is to convert the square wave to a sawtooth, which does provide the whole harmonic family. This can readily be achieved by differentiating the square wave to get alternate up-and-down "spikes," then "losing" one spike by short-circuiting it or blocking through a diode, and finally using the remaining one as a charging pulse for a capacitor discharge circuit (Fig. 8-10). Controlling the tone generation in this sequence enables frequency and amplitude control to be effected on the square wave before it is converted to the sawtooth form. Then the sawtooth will vary in amplitude and frequency in exactly the same way as the square wave from which it was derived, and formants can be applied to the sawtooth output to get still further variations in timbre or tone color.

These are just some of the things that can be done. Such a synthesizer can be programmed to give station signals automatically, to make all kinds of other sound—in fact, the possibilities are endless.

Chapter 9

Loudspeaker Systems

The end result of an audio system is to reproduce sound waves in space, with acoustic properties peculiar to each individual application. All the fluctuating amplitudes and varying frequency content in the audio signal must now be transformed into a related continuum that possesses three-dimensional properties and directions of wave movement in that space. That's why the choice of an appropriate loudspeaker system is of vital importance to the overall result. All that has been done in the electronic part of the system now relies on the loudspeaker system to successfully propagate the program into the space where it will be heard.

INDOOR/OUTDOOR TYPES

This is probably the most dramatic distinction between speaker types. Every indoor installation must "live" with the acoustic characteristics of the room where it will operate. So we will deal with those problems at greater length. An outdoor installation is simpler, due to the absence of this problem, but it does have to withstand weather. Also, in general, it has to serve larger areas without help from reverberation—the containment of the sound.

Outdoor speakers need the maximum efficiency economically feasible, and it is desirable that they be directional to a degree so that sound is not wasted in directions where it is not needed or wanted. For these reasons, horns have become almost universal for outdoor installations: they provide natural weather protection; they are relatively high in efficiency; and they have a degree of directionality dependent on the constants in their design.

Horns come in all sizes and shapes, and are made of a variety of materials, metallic and plastic. Most of the smaller ones, suitable for paging, outdoor intercom, or low-level distribution, come with the driver unit as an integral part. The larger ones have removable drivers so that power can be increased, where necessary, by fitting a more powerful driver unit.

Metal horns are, in general, more rigid, but they do add a "tinny" quality to the reproduced sound. For many outdoor installations this is quite unimportant, as the sound is quite intelligible, which is the main requirement. Horns made of

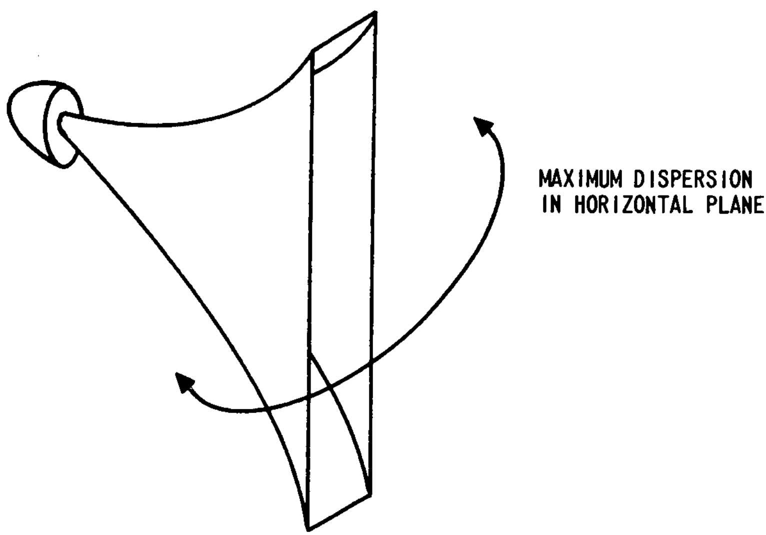


Fig. 9-1. A horn with a rectangular mouth set with its longer dimension vertically, provides wider dispersion in the horizontal plane.

various plastic substances avoid the "tinny" sound, but may still give a trumpet-like effect, which is also quite acceptable in most outdoor installations. The directivity pattern depends more on the internal design of the horn that on its shape, as observed externally. A simple horn, with uniform development of the wave down its length, leading to a rectangular mouth opening, gives the widest dispersion across the narrower mouth dimension (Fig. 9-1). However, some directional horns are designed with changes inside the development that make it possible for the long dimension of the mouth to correspond with the direction of wider dispersion, which is psychologically what one expects.

Horn size relates not only to power-handling capacity; even more directly it relates to low-frequency cut-off. Every horn has a low-frequency cut-off below which it will not operate. If appreciable power at any frequency below this is fed to the unit, it may result in damage, and it will not radiate appreciable sound at such frequencies. The smaller the horn, the higher the cut-off frequency. A wise precaution in any system using horns is to install a bass cut that prevents excessive signal reaching the horns at frequencies below cut-off. The simplest method is a simple capacitor in series with the output, which may be applied either one in each speaker connection or one for the whole system (where all horns are used).

The value of the capacitor (Fig. 9-2) should be calculated to have a reactance equal to that of the load (the horn impedance or combined impedance) at the cut-off frequency. Thus, if all the horns have a cut-off of 200 Hz, and the system consists of eight 16-ohm units, the combined impedance is 2 ohms. A reactance of 2 ohms at 200 Hz requires a capacitor of 400 mfd. Smaller horns may have a cut-off at a higher frequency, such as 800 Hz. If they are each of 45-ohm impedance, and a system uses 15 of them, the combined impedance is 3 ohms. A capacitor to have 3 ohms reactance at 800 Hz must be about 67 mfd. A 50-mfd capacitor of suitable working voltage should serve.

The problem of achieving adequate audience coverage in situations where background noise creates problems is discussed in Chapter 7. This kind of problem is more likely to arise in outdoor than indoor situations. Buildings provide

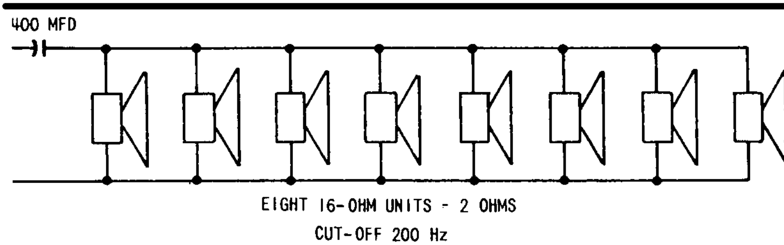


Fig. 9-2. The capacitor in this diagram acts as a simple bass cut-off to protect horns against receiving frequencies below cut-off.

some insulation against outside noises getting in, and the confines of the building enable sound energy to be conserved, rather than escaping to infinity as happens outdoors.

INDOOR ACOUSTICS

The variety of situations that can be encountered indoors is virtually infinite. Studies in acoustics have established certain ideal characteristics, according to building size and the purpose or kind of sound that will make up program. At best these provide only guidelines. The fallibility of such guidelines was dramatized a few years back when the Lincoln Center for the Performing Arts in New York City proved to need quite costly changes, although the guidelines had been followed.

This comes back to the factors we discussed briefly at the beginning of this chapter. The ultimate sound reproduced is a quite complex wave system, propagated through space. In any indoor situation, every listener, or member of the audience, should hear first a direct sound at every component frequency, and then an appropriate amount of reverberation, or reinforcement of the same complex sound. If there is no reverberative reinforcement the sound seems unnatural; if there is too much the direct sound is not strong enough to be intelligible.

Reverberation "happens" whatever you do. Sometimes, as in studios, structural or decorative designs can be used to adjust its amount to better suit the purpose of the sound system. But more often, especially in home high fidelity and stereo installations, the owner just wants to put the system in his room—he's not about to redesign his room to suit the system. So we have to accept the room "as is" and find a way of achieving a good reproduction illusion in that environment.

In the living-room kind of environment, it seems as if nature is "on our side." For smaller rooms the smaller speakers (such as the "bookshelf" variety) seem to do the better job, while for larger rooms the larger enclosures, such as corner horns, are better. But there is more to it than that. And in larger installations other factors enter the picture.

In the home system, room acoustics vary considerably; some rooms are carpeted, wall-to-wall; perhaps the windows are adorned with heavy drapes; and the condition is further augmented with an acoustic tile ceiling and plenty of well-stuffed furniture. Such a room is relatively "dead" acoustically. If you talk, the room gives the impression of being nice and quiet.

At the other end of the spectrum is the "empty" room, or one that sounds that way. The floor may be tile or wood, the walls plaster or paneled, with a hard finish, and the ceiling completely non-absorbent. The furniture may be devoid of upholstery, such as the wrought-iron frame type, with cloth draped over it like a sling. Such a room is acoustically "live." If a number of people are holding simultaneous conversations in such a room, the effect can become quite nerve-wracking. For reproduced sound each kind of room needs its own treatment, for which different kinds of speaker systems are suited.

The reason why speaker size should be suited to room size, apart from the fact that the customer will usually prefer it that way, is that speakers tend to generate sound waves commensurate with their physical size. Of course, every frequency has its corresponding wavelength, and the speaker must have some way of developing waves of each wavelength in the audio spectrum. Large speakers require more space to develop the waves, particularly the lower frequency ones, than do smaller speakers. Consequently, placing a larger speaker in a relatively small room may not give the waves space to develop as the speaker designer intended. This is

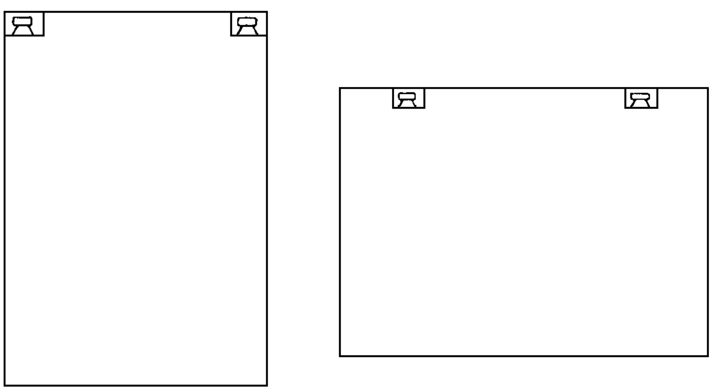


Fig. 9-3. This sketch shows ideal speaker placements in a rectangular room with fairly "dead" acoustic properties.

particularly true of corner horns, and to a lesser degree of bass reflex types.

STEREO SYSTEMS

As most home high fidelity systems are not stereo, we will devote this section directly to achieving the best stereo illusion in different type rooms. For the medium-to-dead type of room, with plenty of absorptive surfaces, the conventional stereo placement is best: two speakers spaced apart so that their separate program content can be clearly heard through most of the room.

In a conventional rectangular room the speakers should be placed either in the corners at opposite extremities of one of the shorter walls, or a little way in from the corners along one of the longer walls (Fig. 9-3). Speaker types can be chosen to suit the situation. In a bigger room, large corner speakers may be best. In a smaller one, perhaps suitable placement for the "bookshelf" type can be found. But even in this type of room, sometimes the ideal placements are not practical. Not all rooms are rectangular—the L-shaped room is very popular in American homes—and doors, windows, and fireplaces are often right where one of the speakers ought to be.

Perhaps an aid to understanding what we try to do in stereo is the concept of an axis of symmetry. In the case of speakers at or near the extremities of one wall, the axis of symmetry bisects that wall and extends across the room, dividing it into two similar halves. In the case of a square room another perfect axis of symmetry would be a diagonal (Fig. 9-4). But in a rectangular room an approximate axis of symmetry could run from corner to corner, a similar diagonal. Putting the speakers on appropriate mid-sides gives a whole range of new possibilities for stereo placement.

Rooms with L shapes or other deviations from the simple rectangular pose a different kind of problem. But usually the simplest concept is to think of the shape as a modified rectangular. The basic rectangle, on which you base your notion of providing stereo, may be part of the whole room, or more than the whole room. For example, if the dining area is a relatively small area added on to the living room, and people will not normally listen to stereo while dining, or will only treat it as incidental music if they do, the best approach is ignore the existence of the dining area (Fig. 9-5). The fact that some of the sound from one side, more than the other of the stereo, will spill over into the dining area may require a slight balance adjustment of the stereo controls, so the balance seems correct.

On the other hand, some L-shaped rooms could be more accurately approximated as a larger rectangle from which a small area is removed (Fig. 9-6). This is particularly true where the remaining area happens to be kitchen, which is only separated from the living-dining area by a counter-top, rather than a complete wall (Fig. 9-7). In either of these cases, it would be easier to consider that you are aiming to serve the entire, large rectangular area with stereo.

Where the room is far more live, as in the average recreation room, without carpets, drapes (heavy ones, at least), or acoustic tile ceiling, or with only some of these, conventional treatment as suggested above may not give a very good stereo illusion because there is too much echo. In the dead rooms

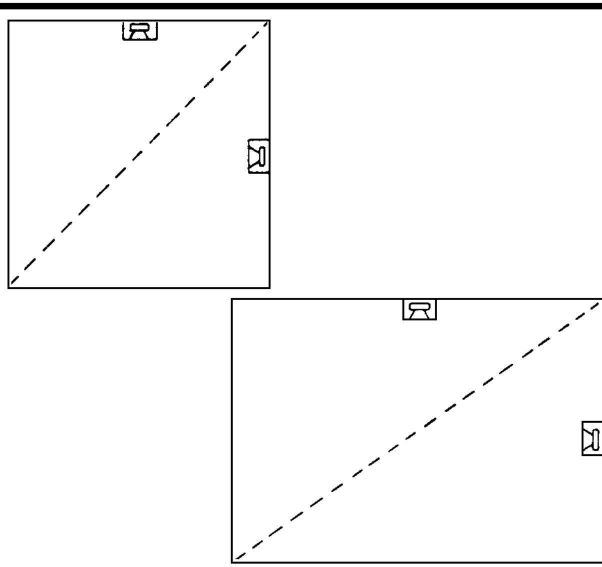


Fig. 9-4. A diagonal axis of symmetry may be used to determine stereo speaker placement, left in a square room, right in a rectangular one.

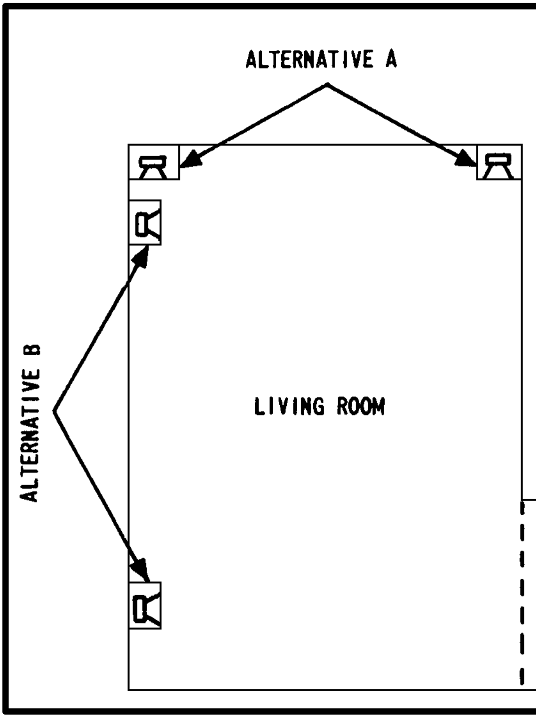


Fig. 9-5. Where the dining area is small in an L-shaped room, its existence may be ignored in considering speaker placement for the living-room area by adjusting the electrical balance to compensate for sound absorption by the dining area.

the aim is to serve the entire listening area with sound direct from both loudspeakers (with the possible exception of spill-over areas). If sound doesn't reach the listener directly, it's apt to be lost, so sitting in a position where only one speaker can be heard will lose the stereo illusion. But in the live room, reflections tend to destroy the illusion if this approach is used. So the technique in a live room is to utilize the reflections, since you cannot easily get rid of them. One way to do this is to use a cabinet-type stereo with speakers on the ends of the cabinet (Fig. 9-8). This relies on reflections from the walls to get the apparent separation, rather than allowing the sound bounced off the walls to destroy it.

Another technique that works well in this kind of room, although it is not confined to this type, is the so-called dipole or planar-type speaker. This must be used quite differently from the other type. All the speakers we have discussed to this point radiate what is basically a sound pressure wave, and the point where this wave enters the room, (or a reflection of it, in the case of Fig. 9-8) identifies the sound

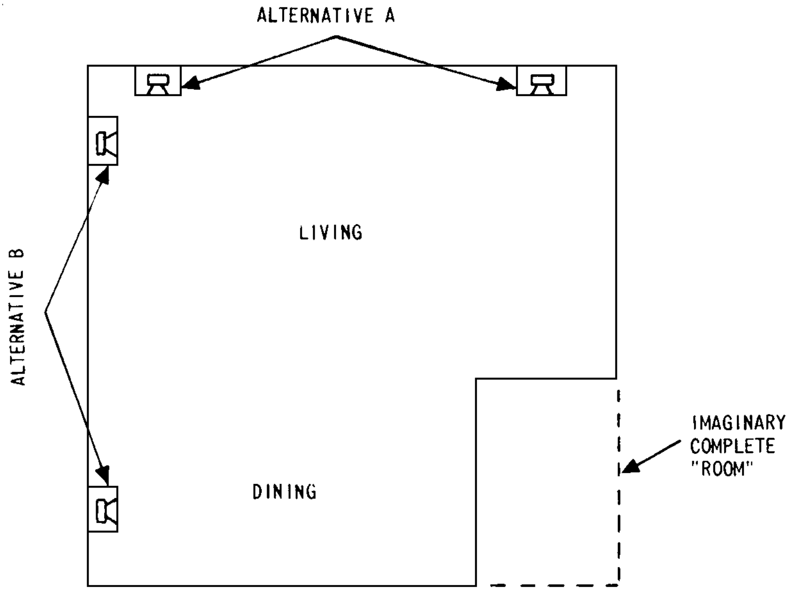


Fig. 9-6. Some L-shaped rooms approximate more closely a large room with a piece missing. In this case the placement is aimed at covering the whole of the larger, theoretical area.

source for each channel. The dipole speaker doesn't work in the same way. The back is completely open so that when the front pushes a sound compression out, the back sucks a sound rarefaction in. Listening to one of the speakers on monophonic program, as you walk around it, the back and front do not sound too different from conventional speakers. But when you get edge-on, the location of the speaker suddenly seems to vanish (Fig. 9-9). You suddenly get the impression that the thing you're looking on as a speaker isn't working, and the sound must really be coming from somewhere else.

If two of these are used on stereo, one for each channel, and placed in a manner somewhat like that shown in Fig. 9-10, a completely new type of listening situation results. Now, in either end of the room, the stereo illusion is as good as it would be with the other type, similarly placed, but facing only toward you. If you used two for each channel, back-to-back, in a highly reflective room, you'd get too much reflection from the ones facing away from you via the far end of the room. But with dipole speakers placed like that in Fig. 9-10, not only do you get a good stereo illusion in both ends of the room, it is also good immediately between the speakers, although you would not identify the sources of sound with the speakers. However, you do get the separation, because of the interaction between the different program content in the two channels.

COST AND EFFICIENCY

Having covered the general methods of providing stereo for different kinds of rooms, some mention should be made about cost and quality. And this in turn is related to efficiency, especially when size is taken into consideration. Small speakers can be highly efficient, provided their frequency response isn't very low, since as small speakers become more effective at lower frequencies (bass) they lose efficiency.

Most earlier books on the subject related quality and efficiency to magnet size—which is the most costly single part of a loudspeaker. With the newer ceramic magnets, this

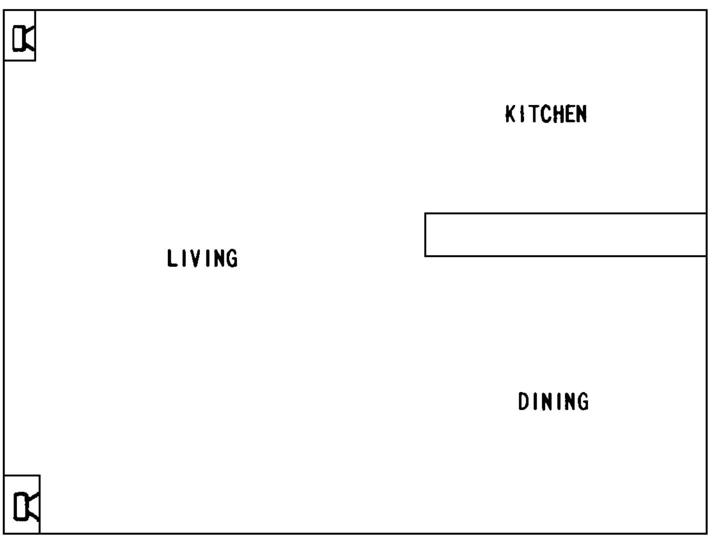


Fig. 9-7. Where kitchen-dining areas have relatively open access between them, the effective room to be served should include the whole of the larger rectangle.

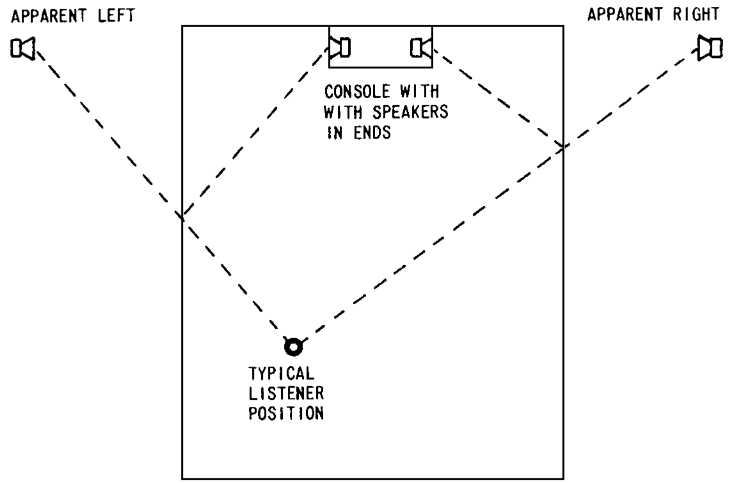


Fig. 9-8. A console type package system provides good stereo in a highly reflective, "live" room, with the speakers in the ends facing out. This utilizes the reflections (rather than treating them as "unwanted") to achieve a greater apparent separation.

relationship is not quite so certain, because the magnet is no longer the vital cost element. And even if magnet size is considered an index of efficiency, this is true only up to a certain size, when other limitations stop a further rise in efficiency with increasing magnet size.

Obviously, if magnet size were the only factor in determining efficiency, this could rise no higher than 100%. After a magnet size sufficient to achieve 50% is reached, further improvement in efficiency would not be commensurate with the increase in magnet size and cost needed to achieve it. But when an objective, such as packing a speaker with good bass response into a small box, is taken into account, a further basic limitation on efficiency is imposed, so that the theoretical ultimate might be 20% instead of 100% (which isn't possible, in fact, because there is no solid substance as light as air from which to construct a diaphragm). Taking all these factors into account, a reasonable "high-efficiency" speaker—one of the larger variety—might run about 25% efficient. A low-efficiency speaker, one of the bookshelf variety, is more likely to run about 2% efficient.

Efficiency is not necessarily related to quality. This is a matter of how well the individual design is implemented. Usually, within the other limitations, improving efficiency within a given type goes along with improved quality, and also higher cost. But technology is advancing so rapidly that a low-cost, well-designed speaker of recent manufacture may be better than older ones costing far more, and on which possibly more design hours were spent!

Speakers for indoor applications, other than high-fidelity or stereo in the home, are made to more competitive economic standards. Unless difficult conditions make it essential to get the maximum acoustic power, requiring a high electrical power with highly efficient speakers, using speakers slightly less efficient can save considerably on cost, and a little more electrical power can be obtained more cheaply than the cost of upgrading all the speakers in the system. Speakers for paging and intercom systems are usually smaller than for the same environment in other applications, with a restricted frequency range and higher efficiency.

MULTIWAY SYSTEMS

Chapter 6 covers various electronic aspects of multiway systems, such as whether to divide the frequencies before or after power amplification. As related to loudspeaker systems, some more basic factors should be considered, pro and con to making such divisions, and how sharply the frequencies should be divided.

Just as there are two philosophies in overall speaker design (one which aims to make the unit as efficient as possible, with the argument that a high efficiency maintained over the frequency range must have a fairly uniform response, by the very nature of it; and the other which sacrifices efficiency for a reduction in size) so there are two philosophies about covering the entire audio frequency range. One of these says that the best speaker is designed to be highly efficient over the maximum possible range. The other says that it is easier to achieve a consistently high and uniform response over a narrower range, so separate units are designed to cover specific parts of the audio spectrum, then frequency division is employed to see that each handles its own part.

From there a further division occurs. If the units are all

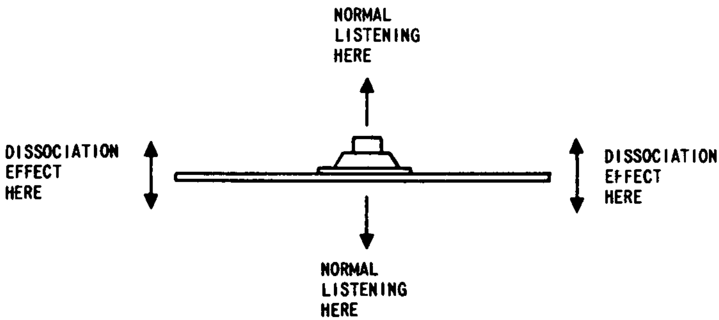


Fig. 9-9. Illustrated in this sketch is another kind of loudspeaker, which gives "dipole" radiation of sound. Used on single-channel sound, the listening effect varies with position.

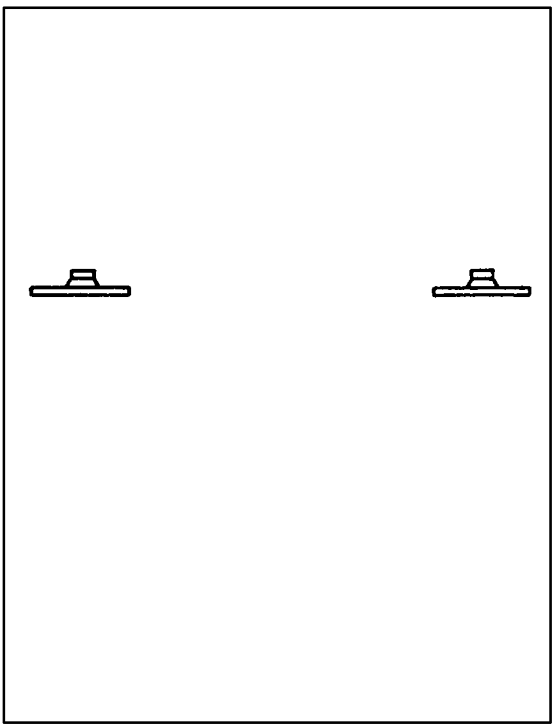


Fig. 9-10. This stereo situation employs two open-backed, or dipole-radiator type loudspeakers.

cone types, but different sizes suited for their respective frequency ranges, then no sudden deterioration of performance occurs, beyond the range for which they are basically used. Thus, the largest unit may respond from 20 Hz to 200 Hz with extremely good uniformity, and above that it will show only a little loss by, say 400 Hz. The mid-range unit may respond extremely well from 200 to 2,000 Hz, but with no serious deterioration down to 100 Hz and up to 4,000 Hz. And the tweeter may respond extremely well from 2,000 to 20,000 Hz or higher, with no serious deterioration down to 1,000 Hz. These three units can be combined with only 6 db/octave crossovers (3 db/octave at the crossover frequency), whether this is achieved before or after power amplification, and the response will blend together quite well.

Many multiway system designers like the superior efficiency of horns. A horn is actually a matching device that works like a transformer to match the moving diaphragm to the ultimate air wave formed at the mouth of the horn, which the cone type can never do. So, over the range for which it is effective a horn speaker is more efficient, and can give a smoother response than the cone type.

It should not be assumed that the horn type is better, per se. Achieving this ideal requires careful design and close precision production. Possibly, equal attention to these aspects can achieve at least equal performance from the cone-type speaker. But audio people tend to be idealists and, if they believe the horn is inherently better, they go that way, regardless.

A disadvantage of the horn type is that when it fails to maintain its uniformity, it does so relatively suddenly. At the low-frequency end a horn has a cut-off frequency below which the horn quite abruptly fails to function as a horn. And at the high-frequency end any factors at the throat or in the horn development that cause deterioration there usually do so quite emphatically, with serious dips and peaks in the resulting response. So a multiway system using one or more horns (some systems combine cone units for some frequency ranges with horn speakers for others) must employ relatively sharp crossover filters so that very little energy is applied to the horn beyond its useful range of frequencies. Crossover filters should be complementary, or the overall response will not be correct. So making the filters sharp for the horns means that filters used for accompanying cone-type units must be equally sharp.

There are two disadvantages to high-slope crossovers: one is electrical and the other acoustical. Electrically, the sharper crossovers are more dependent for precisely correct response and on correct termination than are the simpler types. With the electronic crossover, of which few, if any, high-slope versions are currently available, this can be accurately built into the unit with the use of selected precision components. With the electrical crossover applied at the output, between a single power amplifier and multiple speaker units, the filters are terminated by speaker unit impedances, which are not simple resistances. So the simpler crossovers more easily achieve the responses they are intended to give.

Acoustically, the effect of phase response can be important. Tests have been made with subjective listening that suggest human hearing is not sensitive to phase differences so long as relative levels are not seriously modified (such as by cancellation). There is a fairly obvious reason for this: Acoustic environment produces quite a variety of phase shifts, but the basic character of the sound is still interpreted in spite of the environment, provided this is not severe enough to cause what may be recognized as coloration.

But the phase shifts thus considered are fairly small. If the changes are equivalent to sending the sound through a constricted, resonant pipe, it sounds like it. In a network that provides an ultimate roll-off of 24 db/octave, the phase difference between the two outputs at every frequency (which is most important at crossover) is 360°, or a complete wave. Although the two waves are in phase with each other at that frequency, they are a whole period apart. And in any composite signal most of that 360° phase shift takes place within a couple of octaves. Thus, a shift of a whole wave occurs between a fundamental and its harmonics. And if the system is more than two-way the effect of the various crossovers is additive. A sweep through the frequency range with a sinusoidal tone may give the impression that the response is extremely uniform—as it may well be—but the handling of this system applied to transient program signals may be another thing altogether.

These comments applied to multiway systems on a frequency-division basis do not apply to systems that use a number of similar units to achieve the same or different objectives. For example, a group of units that would not handle the lower frequencies individually may do so easily when they are mounted on a common baffle board, or in the same cabinet, and driven in unison. Consequently, a group of units will handle a wider range of frequencies than each unit can individually. And the beam-type unit, which employs a row of units to make a line radiator (discussed in Chapter 3), applies a combination of units to a different objective—the directing of sound where it is wanted and avoiding sending it where it is not wanted.

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