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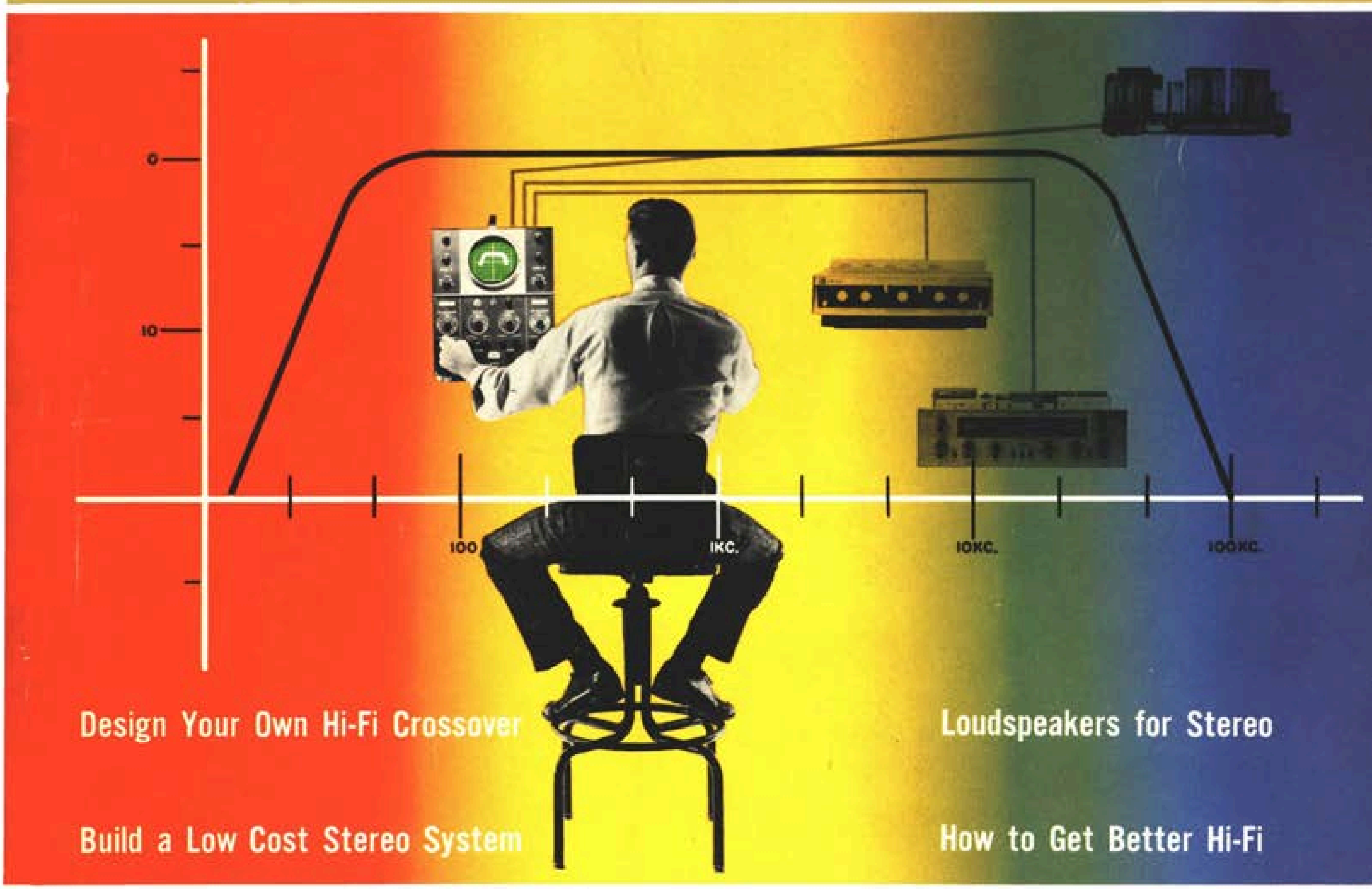

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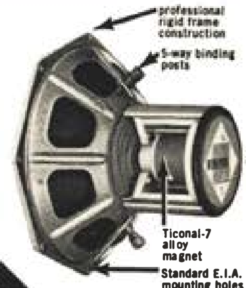

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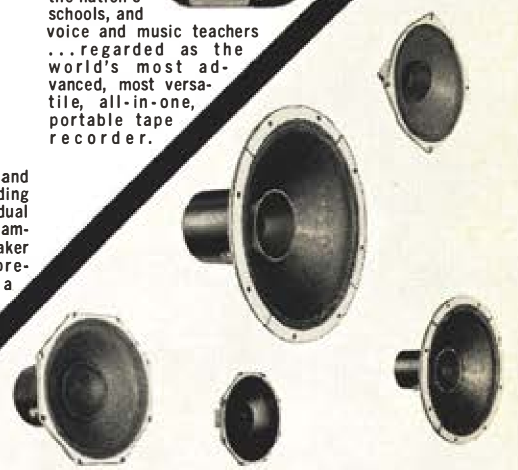

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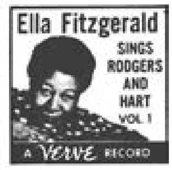
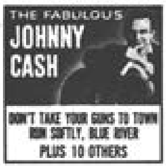


50. Where or When, Manhattan, 10 more

19. No Other Love, Our Love, 10 more



49. That's All Over, One More Ride, etc.

40. "Hallelujah", "Finlandia", etc.

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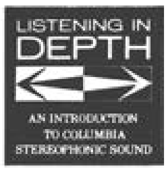

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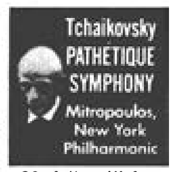
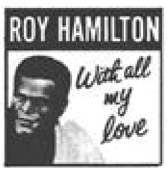
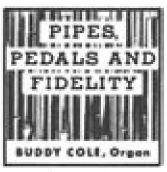


30. A "must" for any record library



9. Always, Please, Speak Low, 9 more

11. Berlioz' most popular work



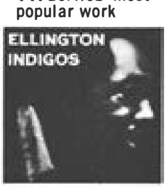
22. Organist Cole plays 11 hit tunes

3. Didn't It Rain, God Is Real, etc.

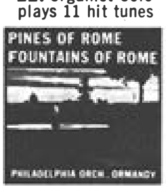
29. High-spirited, gay symphonies

20. Come to Me, Long Ago, 10 more

28. Brahms' most beloved symphony



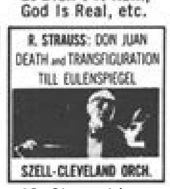
31. Solitude, Autumn Leaves, etc.



18. Two electrifying tone poems

42. Body and Soul, I Got It Bad, 10 more

33. 11 beautiful, immortal melodies

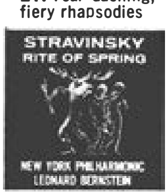


41. Strauss' loveliest tone poems

45. Tico-Tico, Brazil, 10 others

16. Two colorful, exciting scores

21. Four dashing, fiery rhapsodies



36. The ballet that "rocked the world"

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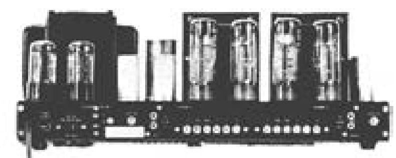


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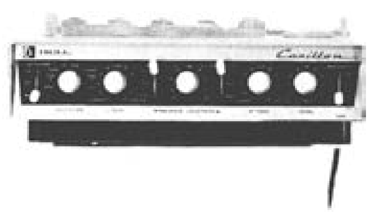

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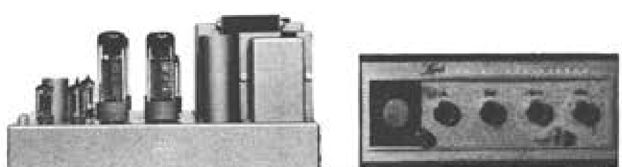

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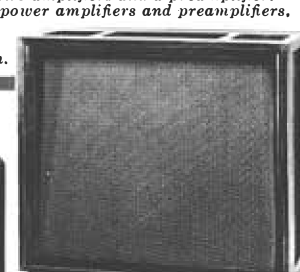

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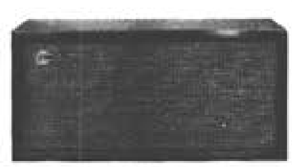

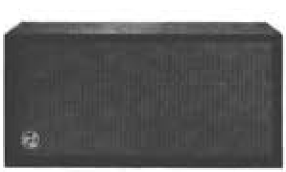

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Richard Strauss: Ein Heldenleben (A Hero's Life), Op. 40. Leopold Ludwig cond. The London Symphony Orch., Hugh Maguire, violin. **LPBR 6038 SDBR 3038**

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Hindemith: Concerto for Violin and Orch. **Mozart:** Concerto for Violin and Orch., No. 3 in G Major (K. 216). Joseph Fuch, violin, with the London Symphony Orch. cond. by Sir Eugene Goossens. **LPBR 6040 SDBR 3040**

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Copland: Billy The Kid, Statements for Orch. Aaron Copland cond. The London Symphony Orch. **LPBR 6015 SDBR 3015**

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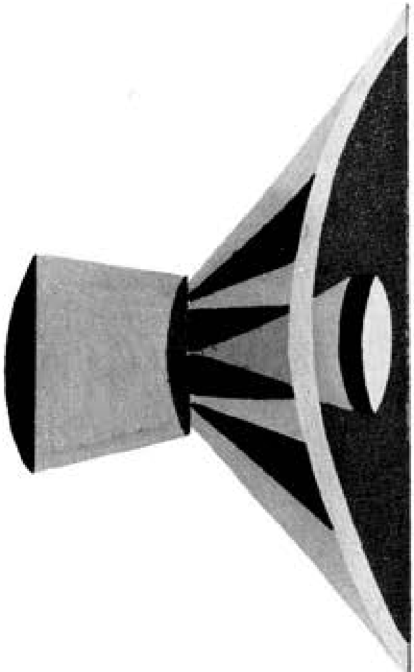
Section I

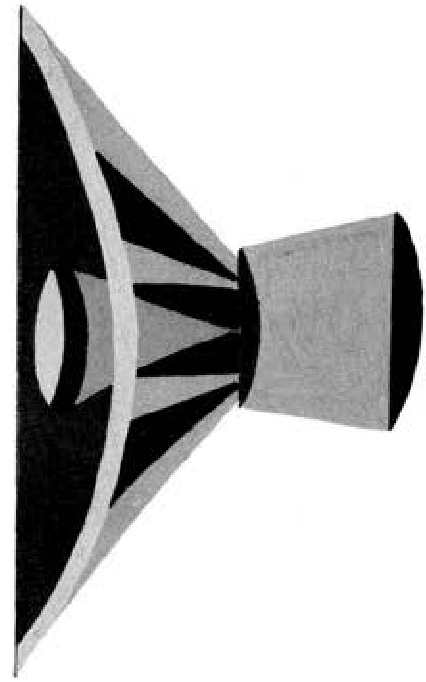
Stereo

The Case for a Third Channel

By **LEONARD FELDMAN**

Results of listening tests when a third channel is added to an already set up stereophonic music system.





AFTER answering consumer questions on stereophonic sound for nearly two years, certain recurring queries served to create a serious doubt as to the validity of some of the accepted axioms of stereophonic reproduction of sound in the home. The problem stems from the nature of the evolution of stereo from its predecessor, binaural sound. Those of us involved in binaural sound, years ago, will always remember dual-headphone listening as a rewarding if somewhat impractical means of achieving spatial realism. We remember, too, that six to eight *inches* between pick-up microphones translated to headphone two-channel listening resulted in virtually perfect spatial visualization on the part of the hearer. The theory advanced then by *Bell Telephone Laboratories* (and there is no reason to believe that human hearing has changed materially in twenty-five years) was that the closely spaced microphones served as individual "extensions" of our two ears, placed in the "best orchestra seat in the house."

There exists even today a hard core of individualists who do the bulk of their two-channel listening *via* headphones. Unfortunately, there is no longer any source material with which to satisfy their binaural craving. Yes, we know that there are nearly a thousand stereo disc titles available (and probably an equal or greater number of stereo tapes), but not one of them was recorded binaurally. The headphone listener is therefore deluding himself—hoping to hear a listening sensation that remains confined to the laboratory.

So much for a few die-hards. The far more alarming question is: are the thousands of newly indoctrinated two-channel stereo listeners equally deluding themselves. Let's examine a typical stereo recording session in detail and see. The first anomaly to strike the observer is the microphone arrangement. If, indeed, the number of channels in the studio pick-up is dictated by the number of ears per person, then the session we are about to witness is intended for outer space consumption, where three-eared Martians dwell. Yes, there are distinctly three microphone channels; a left, a right, and a middle. (There may be more than three actual microphones in a symphonic recording session, but several may be operating to serve only one of the three channels—in effect "compressing" the area of stage-left, stage-right, and center-stage.)

Investigation of this puzzling state of affairs yields two explanations: 1. The recording session is being made monophonically and stereophonically at one and the same time. 2. As long as the center microphone channel is there, why not "blend in" a bit of its output into both the left and right tracks, to eliminate that "hole-in-the-middle" which has so sorely afflicted the listening public? Indeed, why not? Certainly, the "ping-pong" effect originally foisted upon the public to stimulate interest in stereophony is fast disappearing from present-day, sophisticated orchestral recordings. If a fusion or "wall" of sound is the aim, then it would seem to be justifiable to use any technique available in the studio to create that fusion. But what of the home listener? Equipped with two channels, two speakers, two amplifiers, etc., does it follow that he will derive any benefit from the three-channel recording technique? Or, will speaker placement, maladjustment of controls, and room acoustics create a still more objectionable effect—that of *two* orchestras playing at opposite corners of the room. If the latter results (and we have heard numerous installations which do nothing more), then we are spending vast sums of money on superfluous electronic gear when all we really needed was a second speaker. So prevalent is this particular state of confusion that there is not one of us in the field who has not been asked "Is the only requirement for stereo the addition of a second speaker?"

Somewhere between the "ping-pong" effect and "one-channel" stereo lies a happy compromise, which we set out to find. To subject observers to musical listening alone seemed insufficient in the case of three-channel listening. For one thing, the choice of musical selections would, perforce, be arbitrary. The exact technique used in recording would introduce an even greater variable. Our compromise decision, then, was to combine controlled, single-tone experiments with musical auditioning *via* two and three channels.

The Listening Test

Our population sampling, while small compared with some of the monumental works compiled by pioneers in the field, consisted of ten adult listeners. Six males and four females were chosen. Of the six males, two have had some professional dealings with the reproduction of sound. Two were nonprofessionals who had heard stereo before and two had never heard any stereo before the tests. All the ladies had heard stereo, but none were professionally involved in hi-fi. All the subjects were tested first for reasonably equal hearing response in both ears.

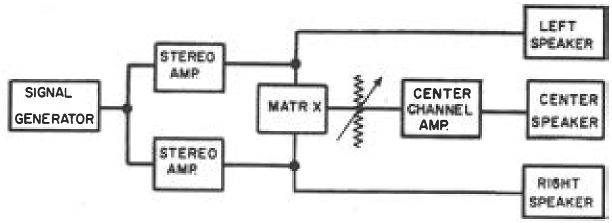


Fig. 1. Block diagram shown here is of the three-channel setup that was employed for the listening tests.

Since our answers were intended to have meaning in terms of home conditions, we elected to conduct the experiments in an average living room rather than in some specially treated sound chamber. In order to eliminate such side effects as "standing waves" and "nulls," each subject was tested from two points in the room. The diagrammatic layout of the test set-up is shown in Fig. 2. In the first series of tests, the same frequency was fed to two loudspeakers and balanced electrically. Unbalance was then introduced, to emphasize either the sound from the left or right speaker. The observer was never told where the emphasis would take place in advance and was asked to indicate when the sound shifted from "center" stage to either the right or the left. This simple objective test was repeated at 10 frequencies, ranging from 50 cycles to 15,000 cycles and at three different levels of intensity. Frankly, after testing the first two observers, we were ready to abandon the project entirely on the basis of such wide divergencies of opinion. As we proceeded with more observers, however, a very definite pattern came into focus. The *average* results of this two-channel test are given in Table 1.

The second series of tests involved the use of a third channel. Much has been written about "three-speaker"

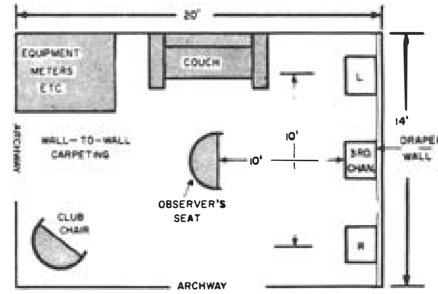


Fig. 2. Room layout in which the two- and three-channel tests were conducted.

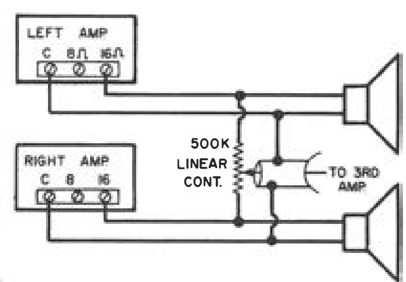


Fig. 3. Matrixing or mixing circuit used.

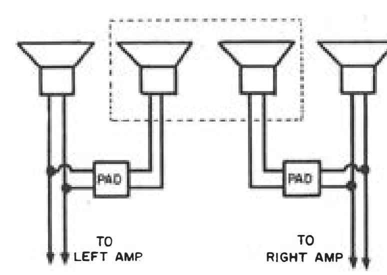


Fig. 4. Creating third channel by paralleling two speakers from left and right channels and positioning them between the outer two. No additional amplifier is needed.

INTENSITY	FREQUENCIES (in cps)									
	50	100	200	400	800	1000	2000	4000	6000	8000 Above
Low	N_I	N_L	5.1 db	3 db	4 db	3 db	3 db	6 db	N_P	N_P N_P
Medium	N_L	7 db	5.3 db	2 db	3.2 db	2 db	3 db	5 db	N_P	N_P N_P
Loud	N_L	5 db	4 db	3 db	3.3 db	2 db	3.5 db	5 db	10 db	N_P N_P

NOTES: Reading of decibels are "average left or right channel emphasis" detected by observers.
N_L = Readings completely random and erroneous, confirming conclusion that low frequencies are essentially non-directional because of long wavelength.
N_P = Readings in these high-frequency ranges, while taken, are deemed inconclusive since errors in judgment of direction were in the majority. At high frequencies, a mere turning of the head will cause the listener to believe that sound has shifted from one side to the other even when sound intensity from one side is as much as 20 db greater than from the other side.

Table 1. Average level differences detected by listeners in two-channel setup.

Table 2. Average level differences detected by listeners in three-channel setup.

INTENSITY	FREQUENCIES (in cps)									
	50	100	200	400	800	1000	2000	4000	6000	8000 Above
Low	N_L	N_L	7 db	8 db	6 db	4 db	6.1 db	7.8 db	8 db	N_P N_P
Medium	N_L	7.3 db	6 db	6.2 db	6 db	4.8 db	7 db	7.3 db	7 db	10 db N_P
Loud	N_L	8 db	6.4 db	6 db	7 db	4.3 db	6 db	8 db	7.7 db	10 db N_P

NOTES: Readings taken in same manner as in Table 1. In addition to increase in one-sided intensity for positive identification of source of sound, it is interesting and somewhat unexpected to note that the range of frequencies over which it becomes possible to determine source of sound is actually extended by the addition of a third channel. It is believed that the center channel serves as a positive "starting point" or mental "anchor point," making phase cancellations less confusing.

stereo, so that perhaps a definition of our "third channel" is in order at this point. We did *not* use a left and right "tweeter" and a center "woofer" channel, for this is still basically two-channel reproduction. We did *not* use one, wide-range center channel and two "end" tweeters or mid-range tweeters. We *did* employ two wide-range systems at the ends and a somewhat less expensive, but nevertheless full-range, system for the center channel. The center channel itself was fed an equal mixture of the left and right channels (the same procedure used in three-channel microphone mixing discussed earlier). The electrical set-up is shown in Fig. 1.

In execution, the second series of

tests was identical to the two-channel test, that is, tones of equal intensity and frequency were fed to the left- and right-speaker systems. A mixture of the left and right outputs was fed to the center speaker at a voltage level equal to one of the end speakers (or, in other words, 3 db lower than the total power of both end speakers). The sound of one channel was increased in intensity until the observer could state with certainty that the sound seemed to be coming from a particular direction. The average results of this second series is shown in Table 2. By comparing equivalent average measurements between the two series of tests, one immediate conclusion can be drawn with respect to three channel. It takes *more* change of emphasis of either left or right channels before the listener can definitely "fix" the source of sound. For example, at medium sound level of 1000-cycle tones, the listener felt a shift from center to either left or right with a change of only 2 db. In the three-channel set-up, a change of 4 db was required before the observer detected the "shift" of location. The conclusion is that it is *far* more difficult to achieve true balance in a two-channel stereo set-up than in three channel. In fact, we seriously doubt if balance can be maintained or adjusted for a given point in the room without instruments. This does not mean that under musical conditions stereo illusion is lost. It does mean, though, that in two-channel stereo the vocalist or soloist originally at center stage may alternately appear to be at stage-right or stage-left by a mere shifting of the listener in his chair. This did, in fact, take place in the musical tests which followed.

Musical Corroboration

To translate the tabulations into further meaningful conclusions, we selected vocal and instrumental discs and tapes and subjected the listeners to them. In all ten cases, the listeners preferred having the third channel in use in this form of musical presentation. (Of course, the listeners were never told whether the third channel was in or out and *over-all* sound level at the listener's seat was maintained constant. As the third channel was faded in, end channels were reduced proportionately, to maintain equal *total* sound at all times.) In the case of solo or vocal music surrounded by an orchestral background, the reason given for the preference of third channel was essentially "I find that I don't have to concentrate on *where* the soloist is standing." In purely orchestral music, only six out of ten preferred a third channel. Two experienced no perceptible difference between two and three channels and two actually preferred the two-channel arrangement. It is interesting to note that these latter two were the two gentlemen who work professionally in the audio field. It is possible that the extensive stereo listening these men have done recently may be partly responsible for their subjective answers as to preference.

Whence the Third Channel?

If your interest is aroused at this point, and if you would like to draw your own conclusions, it is not difficult to duplicate the set-up we used. The third channel was derived right at the speaker terminals of the left and right channels. The "hot" lead to each of the end speakers was fed to a 500,000-ohm potentiometer (one lead to an end terminal of the potentiometer, the other speaker lead to the other end terminal of the potentiometer). The arm of the pot was then used to feed a third basic power amplifier and speaker. By rotating the shaft of the potentiometer (which had a linear resistance element) half way, equal amounts of left and right signal were picked off at the arm of the pot. A schematic of the arrangement is shown in Fig. 3. All grounds (from left and right speakers and shield of cable going to third amplifier) were tied together. It is extremely important that the third speaker be phased properly with respect to the other two. If increasing the level control on the third amplifier seems to *reduce* the total level of sound in the room, you can be certain the third speaker is phased incorrectly and a simple reversal of the leads to that speaker will correct the situation. It goes without saying that the left and right speakers should also be phased properly.

Several preamplifiers currently available for stereo set-ups feature provisions for a third power amplifier connection. These include *Madison Fielding* Model 340, *Scott* Model 130, and *Lafayette* Model KT-600. In these preamplifiers, the necessary "blending" is accomplished electronically, providing the user with a "mixed" left and right signal for just this application. Undoubtedly, others will be marketed.

Still a third method of achieving three channels suggests itself. In this last method, no third amplifier is required, but two more wide-range speakers are involved. The set-up is shown in Fig. 4. Here, a third wide-range speaker is paralleled across the left speaker and a fourth speaker is paralleled across the right speaker. Physically, however, the two extra speakers are mounted in one enclosure which is positioned between the other two enclosures. If you have a pair of speakers currently doing stereo duty in another room, this might be the most inexpensive way to perform the three-channel experiment without additional expenditure until you decide whether or not you like the effect.

As for our own situation, there will be no rest until I replace the speaker system I "temporarily" stole from our bedroom and am unwilling to return—I happen to have decided in favor of three-channel stereo in the living room. For the time being, our bedroom TV set will have to subsist on its own 4" low-fi unbaffled "squeaker."

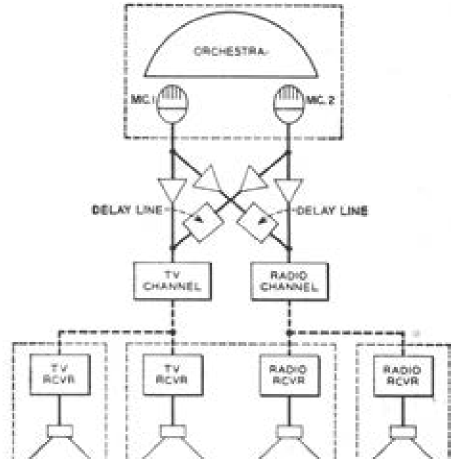
Compatible Stereo Broadcasts

New Bell Labs system uses a time-delay effect for stereo.

MUCH present stereo broadcasting is not compatible in that the listener to one channel does not receive the entire program information. Now this single-channel problem may be eliminated without affecting the stereo listener, through the use of a new "compatibility" circuit which has been developed by F. K. Becker of *Bell Telephone Laboratories.* The circuit depends for success on a psycho-acoustic phenomenon known as the "precedence effect." This effect operates in such a manner that when a single sound is reproduced through two separate loudspeakers, but is delayed several milliseconds in one, the listener will hear the sound as if it came only from the speaker from which he heard it first. He will judge the second loudspeaker to be silent.

In the new development, circuits are cross-coupled (see diagram below) through two 10-millisecond delay lines, each with its own amplifier. Signals from the left mike are sent directly to the left speaker, while the same signal is delayed 10 msec. before reaching the right-hand speaker. The stereo listener will hear the sound as if it came only from the left speaker because of the precedence effect. Also, sound from the right mike goes direct to the right speaker but is delayed before reaching the left speaker, and is therefore unheard. The stereo listener thus localizes the sound in such a way that a full stereo effect is produced. However, monophonic reception is completely compatible with this, since a listener to each single channel hears the total sound from both mikes in a balanced reproduction. The slight delay produces no echo and does not affect reception at all, according to subjective tests.

Basic setup employed in compatible system.



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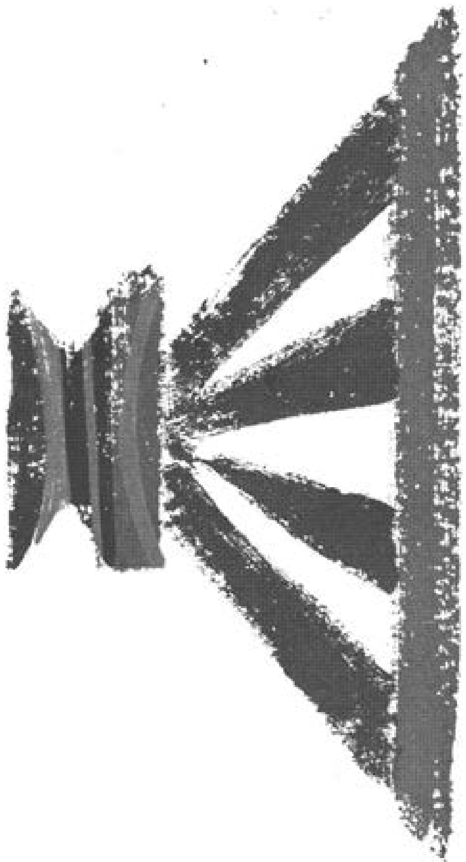
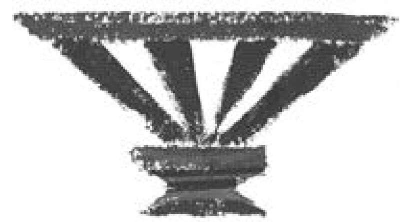


Phantom Channel for Stereo



By **HERMAN BURSTEIN**

Simple methods that may be used to obtain a third or center channel to eliminate that "hole-in-the-middle."

ONE of the major problems of two-channel stereo is the apparent absence of sound in the space between the two speakers. The more widely separated the speakers, which augments the sense of spaciousness, the more severe is the "hole-in-the-center" effect. Therefore an increasing measure of attention has been directed toward filling this seeming void. One way of doing this is to use three-channel stereo. However, since three-channel program sources are not available to the public, the "fill" must be derived from two-channel material.

A simple way of filling in the center is to move the left and right speakers closer together. If corner speakers are employed, this is difficult or impossible to do. Even if wall-type speaker systems are involved, problems of furniture arrangement may interfere with bringing them together.

Another expedient used on occasion is that of a dummy speaker, connected to nothing at all, which is situated between the left and right speaker systems. Sight and sound go together much as taste and smell do and the visual presence of a speaker system in the center can lead the mind to conclude that it hears sound from this region. A third technique is to conceal the two stereo speakers behind a curtain, so that one is not conscious of speaker separation, enabling the two sounds to blend together in the mind.

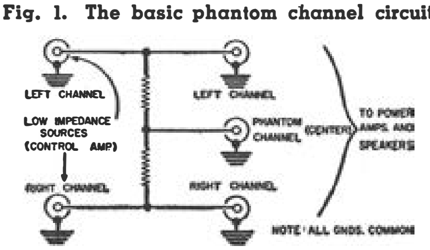
What is apt to be a more satisfying approach is the "phantom-channel" technique, where signals from the left and right channels are combined and fed to a central speaker. The phantom-channel signal can be derived either before or after the power amplifier. If after, one saves the cost of a third power amplifier.

One early proponent of the phantom channel as a solution to the problem of hole-in-the-center was Paul W. Klipsch of Klipsch speaker fame. His method combines the left and right signals prior to the power amplifier, so that a third power amplifier as well as a third (central) speaker is required.

Fig. 1 shows the essence of Klipsch's method while Fig. 2 shows the refined circuit, including values of the resistive network, chosen to provide adequate isolation (minimization of crosstalk) between channels and the proper level of the middle channel relative to the other two. In Fig. 2 the mixing function is performed by the 33,000-ohm resistors, while the 82,000- and 220,000-ohm units attenuate the signals to the left and right channels. Klipsch in his experiments used a 5000-ohm source in the left and right channels, so that the 33,000-ohm resistors provide slightly better than 20 db attenuation of crosstalk between channels. If a cathode-follower source were employed, typically with an output impedance of about 500 to 700 ohms or less, crosstalk attenuation would be about 40 db. Generally, 20 db attenuation is considered adequate for stereo.

A vital feature of Klipsch's network is that the level of the central channel is *higher* than that of each of the end channels. To be specific, the middle channel is designed to be 3 db higher than each of the others or, in other words, equal to the *combined* level of the left and right channels. Of course it is assumed that the power amplifiers and speakers for each channel are identical, resulting in an acoustic output which is greater for the center speaker than for the flanking ones.

Fig. 1. The basic phantom channel circuit.



Klipsch states in this matter: "Some guessing was done as to the level to be fed the center channel and the guesses were all wrong. Experiment led to better thinking and a theoretical basis was arrived at and corroborated. . . . My experiments resulted in a workable system with the center channel a half-and-half mixture of the two sound tracks (from a tape) and the flanking channels using corner speakers fed from the two sound tracks with 3 db attenuation relative to the center channel. . . . When success finally was achieved in balance, a surprise occurred: the center channel was perfectly real, and not just a simulated effect to fill up a hole in space. Sounds remembered as arising in the center of the stage occurred there; one ceased to hear sounds from the three speakers, and actually sensed a spread across the curtain of sound."[1]

Contrary to Klipsch's findings, others have recommended that the center channel be substantially *lower* than the end channels. Thus it has been advised that "the gain of the center channel amplifier should be adjusted so that the sound from the center speaker is just audible."[2] Another has recommended that the center channel should be below the end channels.[3]

Possibly the discrepancy in views as to the proper level for the center channel relative to the left and right speakers may be explained by variations in speaker location or by the nature of the stereo material used for the pertinent experiments. For example, if very wide microphone spacing were employed in the original recording, it might be desirable—in the sense of greatest listening satisfaction—to compensate for the abnormal stereo effect by elevating the volume of the center channel. Or if the left and right speakers are spaced a relatively great distance apart, so as to form an angle of more than about 50° with the listener, it may be desirable to accentuate the center channel. On the other hand, the narrower the microphone and/or speaker spacing, the lower may be the level of the center channel for optimum listening results.

All-in-all, it appears that the stereophile should not commit himself to a fixed relationship between the center channel and the other two but should leave the situation open to experimentation. He should have a handy means of varying the relative level of the center channel, for example by a readily accessible potentiometer in the control amplifier or power amplifier, or by a pad in the leads to the speaker system. (The last approach may raise problems because a pad intervening between the power amplifier and speaker system may deteriorate speaker damping to the point where hangover and/or ringing become apparent.)

As shown in Fig. 3, Klipsch has suggested a means for eliminating the third power amplifier. Here, however, the level of the middle speaker would be 6 db down with respect to each of the flanking speakers instead of 3 db

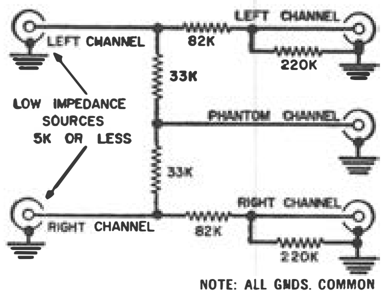


Fig. 2. More elaborate version of Fig. 1.

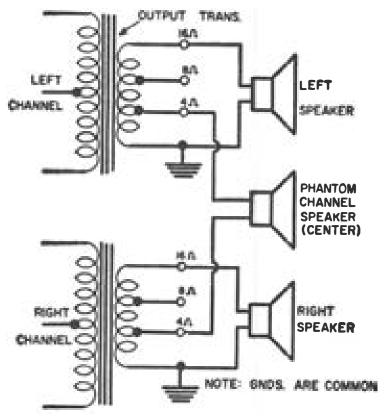


Fig. 3. Obtaining a third channel at the outputs of the power amplifiers employed.

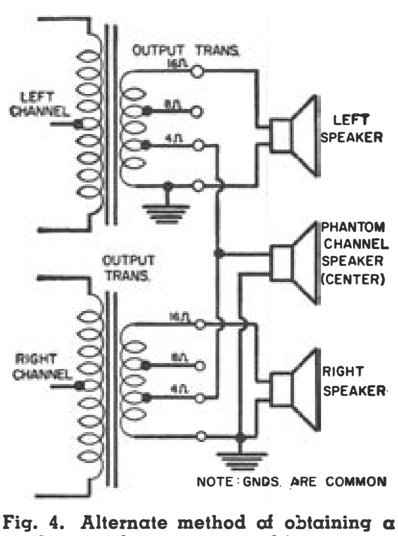
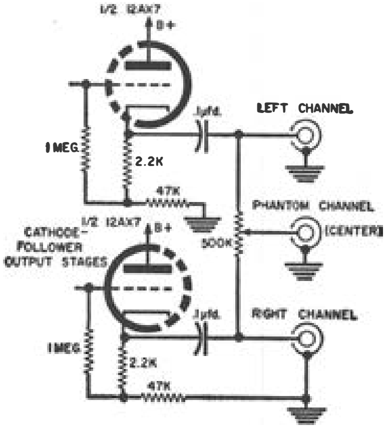


Fig. 4. Alternate method of obtaining a third channel at power amplifier outputs.

Fig. 5. Circuit used in Madison Fielding Series 340 stereophonic control amplifier.



higher in level. This assumes speakers of equal efficiency on each channel. One could change the situation by using a center speaker of greater or lower efficiency than the others.

The method of Fig. 3 introduces a question of crosstalk. But this seems to be not at all serious. The writer, pursuing Klipsch's method, introduced an 8-ohm resistor, in lieu of a speaker, between the 4-ohm terminals on a typical stereo amplifier. He found that crosstalk attenuation was 28 db.

Another question raised by Fig. 3 is that of signal cancellation. Assume there are equal signals in the left and right channels, as can happen, depending upon microphone techniques. In such a case, there would be no electrical potential between the 4-ohm terminals (or any other pair of like terminals) and hence no signal. However, in actual stereo source material there are generally differences in amplitude and phase between like signals, so that signal potentials will exist between like terminals.

To the extent that signal cancellation does exist, an alternative technique to prevent such cancellation would be to connect the middle speaker to the outputs in parallel as in Fig. 4. However, this results in considerably greater crosstalk. It was found that crosstalk attenuation dropped from 28 db to 16 db when the hookup of Fig. 4 was employed. When crosstalk attenuation drops much below 20 db, it is apt to interfere with the stereo effect.

At least 3 stereo control amplifiers now on the market incorporate a phantom-channel output. These are the *Madison Fielding* 340, the *H. H. Scott* 130, and the *Lafayette* KT-600.

Fig. 5 reveals the method employed by *Madison Fielding* to derive the phantom channel. The left and right signals are combined by a 500,000-ohm pot connected to the high side of the left and right output jacks. The arm of the pot goes to the phantom channel output jack. The left and right channels have cathode-follower output, with an impedance of about 700 ohms and the linking 500,000-ohm resistance introduces negligible crosstalk, *i.e.*, channel separation is nearly 60 db.

Use of a potentiometer as a combining device, instead of fixed resistors, allows one to balance the relative levels of the left and right signals for purpose of combining them into a center channel. Such balancing may be desirable for a number of reasons. For example, the balance control in the Series 340 might be set to produce different signal levels in each channel due to variations in power amplifier gain or variations in speaker efficiency. A potentiometer enables the user to again obtain equal signals for center channel purposes. To take another example, the listener may wish to obtain *unequal* signals in the center channel, say a dominantly left signal, because the center speaker cannot be located exactly midway between the other two but is situated closer to the left one because of problems of the furniture arrangement.

The mixing technique of Fig. 5 results in the center channel having a relatively high output impedance, so that more than two or three feet of cable cannot safely be used without loss of highs. On the other hand, loss of highs on the center channel might be more of a virtue than a fault. To the extent that the highs are associated with directionality, that is, with left-right orientation, it is just as well for them to disappear from the center.

Fig. 7 shows the mixing technique used by *H. H. Scott* in its Model 130. In basic respects, the method is similar to that of *Madison Fielding.* However, there are two important differences. One, the mixing ratio is fixed instead of variable. This can be compensated, if need be, by adjusting power amplifier gain (most power amplifiers have an input level control) so that equal sound will emanate from the left and right speakers when the balance control of the 130 is at mid-position, thereby producing equal signals in the left and right output jacks and making equal contributions to the phantom channel. The second difference is that the combined signal is not fed directly to the center channel output jack but goes first through a triode having low output impedance due to substantial negative feedback. Thus cable lengths up to 20 feet between the preamp and the power amplifier will introduce no significant loss of highs.

As can well be realized from the preceding discussion, incorporation of a phantom channel adds further complexities to an already complex stereo situation, what with problems of balancing channels, of phasing, of simultaneously controlling the gain of channels with minimum tracking error, etc. Yet at the same time the phantom channel holds forth a satisfactory solution to an important problem, the hole-in-the-center. And in itself the third speaker system enhances the sensation of breadth, one of the essential attributes of stereo.

Furthermore, use of a phantom channel may well be a means of solving the problems of cost and space for many audiophiles contemplating a stereo installation. It may seem contradictory that a stereo setup employing three speaker systems can be less expensive and occupy less space than one having only two speaker systems, but Fig. 6 illustrates how this may come about.

As shown in this diagram, the center speaker system is the expensive and, as generally happens with expensive speakers, the big one. Quite likely it is the one hitherto used by the listener for monophonic reproduction. Instead of having to find the funds, as well as the space, for a matching speaker system, he can employ small, matched, inexpensive speakers for the left and right channels. The center speaker provides the total audio information, particularly the low bass, which is essentially non-directional and not substantially associated with the stereo effect. The flanking speakers, being small and inexpensive, do their relative best in the upper bass, middle, and treble ranges, which are most closely identified with the stereo effect. The cost of the two small speaker systems for the left and right may be considerably less than the cost of matching the monophonic speaker presently owned by the audiophile. Or, if he is starting from scratch, the cost of one large speaker system and two small systems can be appreciably less than the cost of two large ones.

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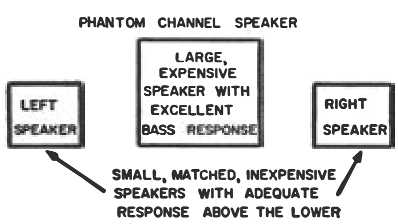
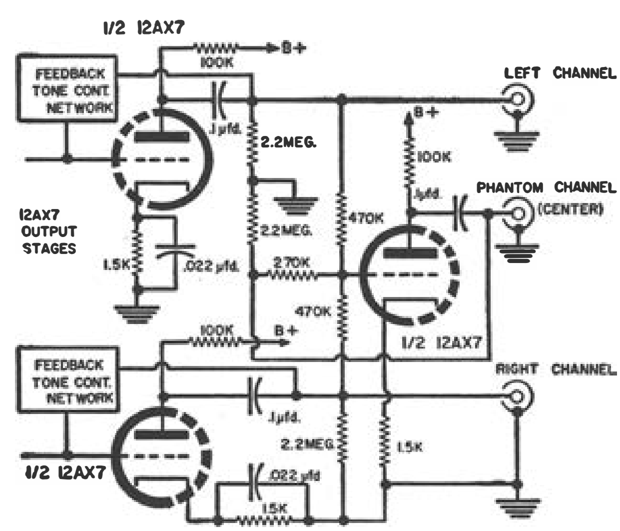
2. *Personal communication from D. R. von Recklinghausen, Chief Research Engineer, H. H. Scott, Inc.*
3. *Personal communication from Leonard Feldman, President, Madison Fielding Corp.*



▲ **Fig. 6. A stereo loudspeaker installation is shown above that incorporates a phantom channel arrangement as described in the text.**

Fig. 7. Here is the circuit that is employed in the H. H. Scott Model 130 stereophonic control unit. ▶

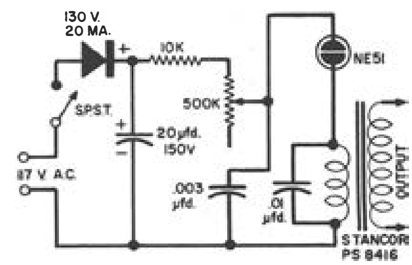


LOW-DISTORTION SINE-WAVE GENERATOR

By HERBERT COHEN
General Transistor Corp.

THE author has been considering the problem of extracting the fundamental frequency from a conventional neon-tube saw-tooth generator. The slow charge time and the extremely rapid discharge time produces a saw-tooth waveform whose usefulness is limited but which contains the fundamental and many harmonics.

However, by parallel tuning the capacitor discharge circuit, a low-distortion sine wave can be obtained. The primary of a Stancor PS8416 power transformer, paralleled by a .01 µfd. capacitor, is used as a high-"Q" tank circuit. The tuned circuit appears as an exceedingly high impedance to the fundamental frequency but practically a short circuit to its harmonics.



With an oscilloscope across the transformer primary, the pot is adjusted to set the oscillator at the LC frequency of transformer and capacitor. As the fundamental approaches the tuned LC frequency, the waveform becomes more and more sinusoidal with an increasing amplitude on approaching resonance.

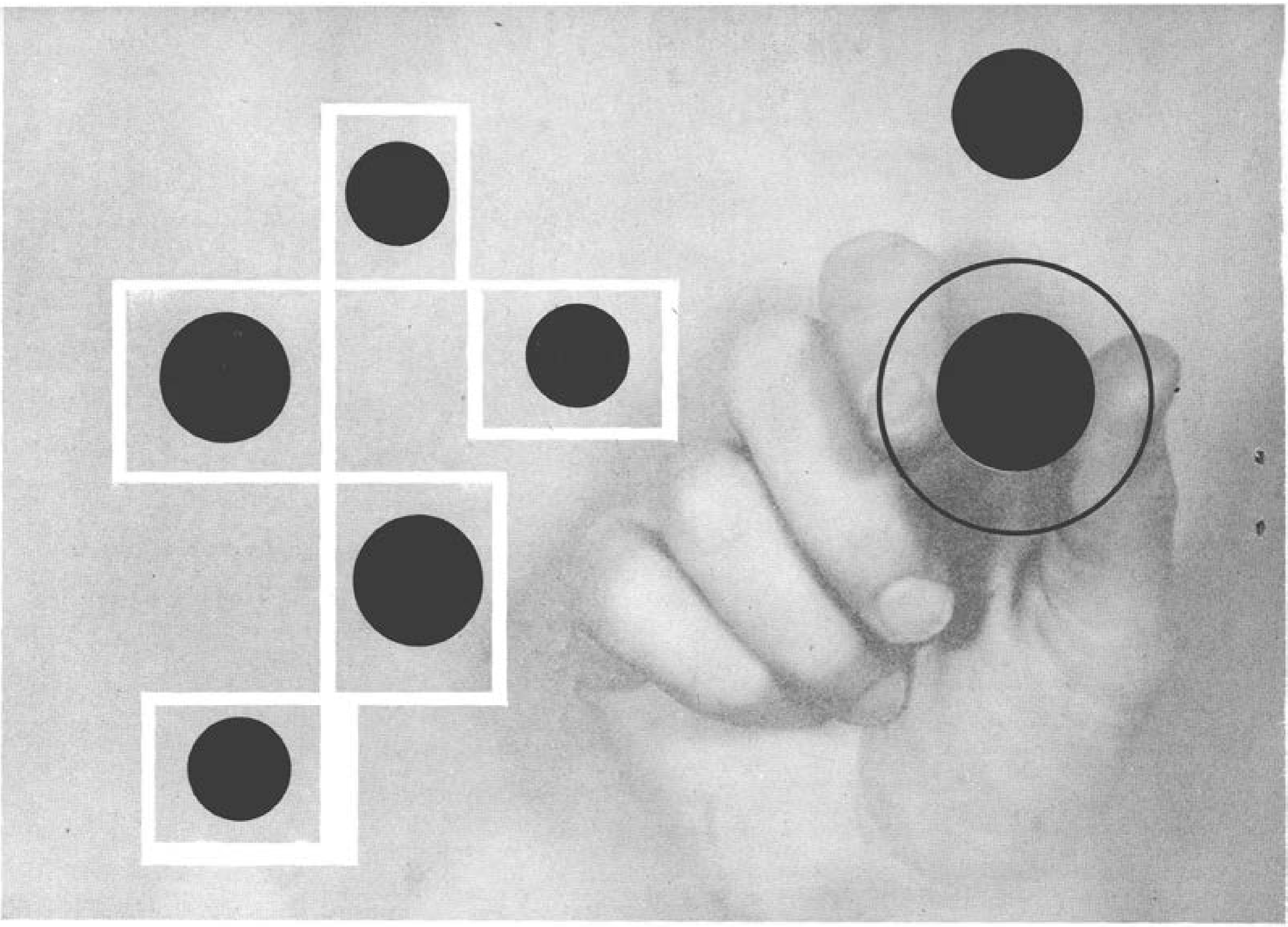
At resonance, the waveform shows less than 3% harmonic distortion as measured with a distortion analyzer. The secondary of the transformer supplies many outputs for low- or high-impedance matching. One problem in this construction is that ground is also a.c. ground. An isolation transformer can be used to eliminate this problem. The stability of this unit is basically determined by the stability of the a.c. line. A 1000-cycle oscillator, constructed by the author and shown in the diagram below, puts out a waveform comparable to the Hartley type.

TRANSISTOR SOCKET HOLDER FROM FAHNESTOCK CLIP

By JOSEPH A. BORSOS

WHILE building my second transistor radio recently and ruining an expensive "p-n-p" transistor when I soldered it in the circuit, I looked around my shack and spotted a Fahnestock clip in my junk box.

Simply straightening the top portion of this spring connector clip and pressing in the wire-holding prong yields a most economical transistor socket holder, which is unusually adapted to the breadboard circuit since it can be mounted with a single screw! It may be necessary to file out the slot a little in order to accommodate some transistor sockets.



STEREO AMPLIFIER CONTROLS-
and what they do

By **HERMAN BURSTEIN**

A good over-all summary of the functions of the various operating controls that are used in a stereo system setup.

THE stereo amplifier has a dual task: to coordinate two program channels in conventional functions, such as selection of input source, gain control, equalization, etc.; and to perform certain functions unique to stereo. We are talking essentially about control amplifiers, although most of the following discussion also pertains to integrated equipment such as a control amplifier combined with a power amplifier or with a tuner.

The stereo amplifiers of various manufacturers differ to some extent in the functions they provide. In choosing among stereo amplifiers, the prospective purchaser must consider what features are apt to meet his needs. An understanding of the controls and functions commonly found in these units will facilitate his choice. The first control discussed is the balance control.

Unique to stereo is the balance control (sometimes called a focus control) permitting the level of either channel to be increased relative to the other by means of a single knob. Fig. 1 shows the basic principle of many balance controls, enabling the level of one channel to increase as the other decreases so that total volume remains nearly constant.

Whereas Fig. 1 employs two ganged potentiometers, it is possible to control balance by means of one pot, as shown in Fig. 2, where the arm causes one channel or the other to be shorted to ground as it moves up or down. The values shown in Fig. 2 will maintain the combined sound level of the two channels reasonably constant as the control is rotated.

The range of action of the balance control varies greatly among stereo amplifiers. In some maximum rotation of the control produces as little as 6 to 10 db difference in level between channels, while in others the difference is as much as 40 db, or even greater. Fig. 1 permits an infinite difference. Fig. 3 shows how the range can be limited by inserting a resistance between each pot and ground. Fig. 2 also permits an infinite difference, while Fig. 4 shows how this circuit can be modified to limit the maximum difference between channels.

The ratio between highly efficient and quite inefficient speakers is roughly 20 to 1, or 13 db. Power amplifier sensitivities differ by as much as 14 db (one amplifier may be driven to 10 watts by .2 volt, while another may require 1 volt). The two sections of a stereo cartridge may differ in output level by 3 or 4 db. The two channels on a stereo disc or tape may differ by several db. All told, it is possible for these differences to add up to a very considerable total, so that a balance control with a range of action approaching 40 db may prove useful, assuming the stereo installation comprises unmatched power amplifiers and unmatched speaker systems.

But if one employs matched power amplifiers and matched speaker systems, the need for a wide-range balance control diminishes. A range of 10 db or less can then be satisfactory. Moreover, the limited-range control has the advantage of a finer vernier, which makes accurate channel balancing easier.

Master Gain Control

Once the stereo system is adjusted for balance between channels, it is important that the listener be able to change total volume without upsetting channel balance. Therefore, most stereo amplifiers contain a master gain control, as in Fig. 5, consisting of two pots operated by a single shaft.

A few stereo amplifiers have, instead, individual gain controls for each channel, but concentrically mounted so that by rotating both together one achieves the effect of a master gain control. It is usual practice, then, to use concentric shafts that lock together when the inner shaft is pushed in slightly.

The change in balance that occurs when rotating the master gain control is called "tracking error." This should be no more than ± 3 db, and preferably no more than ± 1 db. Conventional pots mounted on a single shaft may produce errors much greater than ± 3 db, due to differences in taper between the pots. The tracking error can be kept within suitable bounds by the following means:

1. Employ pots manufactured to close tolerances. This is expensive.
2. Employ conventional pots, selected to form matched pairs with closely similar tapers.
3. Employ pots with taps, as in Fig. 6, to bring the two pots into correspondence at a number of intermediate points between minimum and maximum rotation.
4. Employ stepped controls, as in Fig. 9. Resistor values are usually chosen so that going from one switch position to the next produces a level change of about 2 or 3 db. Tracking error can be kept well under 1 db by this method, however, it is quite expensive and has a limited range of attenuation due to the finite number of switch positions.

Gain controls are best located at an early stage in the amplifier in order to reduce the signal before it can drive a stage hard enough to cause appreciable distortion. It is desirable to have the balance control as well as the gain control at an early stage, thus the master gain and balance controls are often brought together, as shown in Fig. 7.

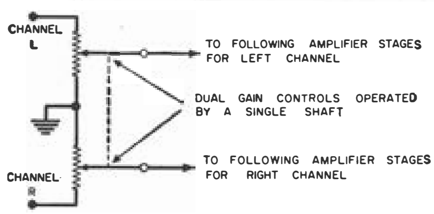


Fig. 1. Here is the circuit arrangement employed in a basic balance control hookup.

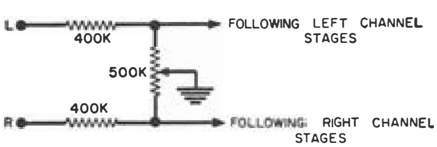
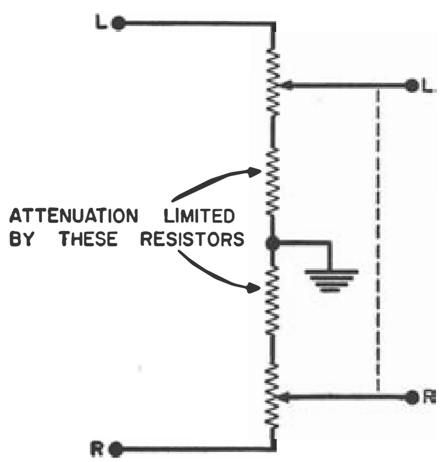


Fig. 2. Single control for balancing.

Fig. 3. A balancing arrangement that employs a limited amount of attenuation.



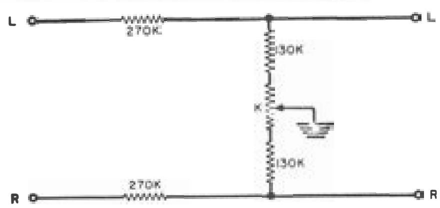


Fig. 4. Single control for channel balancing with limited (about 10 db) attenuation.

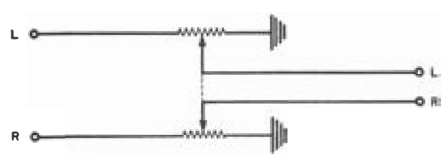


Fig. 5. A typical basic master gain control.

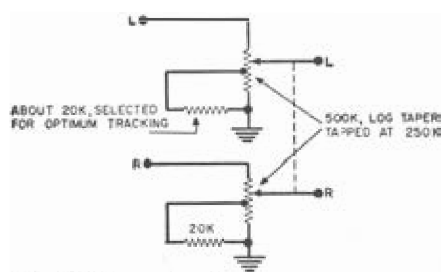


Fig. 6. Tapped controls for good tracking.

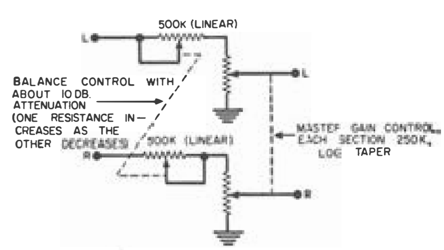
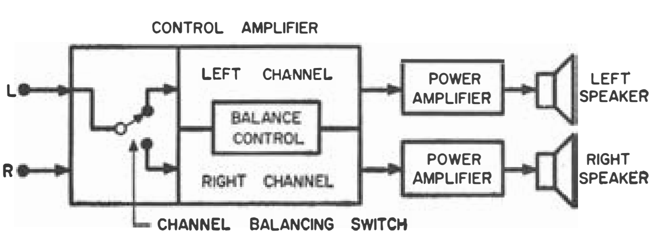


Fig. 7. Combined balance and master gain.

Fig. 8. The system diagram at the right shows hookup that may be employed for channel balancing. See text for operation details.



Tone Controls

It is still a moot question whether separate or ganged bass and treble controls should be incorporated in a stereo amplifier. The ganged control simplifies operation and appearance and, if matched speaker systems are employed, the chances are reasonably good that this type of control will provide enjoyable listening.

Individual controls, however, afford greater flexibility. If different loudspeakers are used for the two channels, different amounts of bass and/or treble correction may be required in each channel to achieve best results. Even when using matched speakers, their location and orientation with respect to the listener may call for varying tonal correction. Differences in the signal for each channel, as in the case of FM-AM stereo, may also call for different bass and/or treble adjustment. Yet with further advances in the art, including the transition from FM-AM stereo broadcasts to FM multiplex ones, increased use of matched speaker systems, and greater uniformity of frequency response between channels on stereo discs and tapes, it may be expected that the present advantages of separate tone controls will probably diminish.

Channel Switching

Stereo amplifiers provide various switching functions to permit balancing channels, listening to monophonic sources, correcting for errors in phasing or channel identification (left *versus* right), etc. Most stereo control amplifiers provide most of the following functions, although few provide all of them.

1. *Reverse Stereo:* This permits the left signal to be fed to the right channel and the right signal to the left channel, as shown in Fig. 10. While accidental reversal of channels on commercial discs and tapes can be expected to drop to zero, there is as yet no stand-

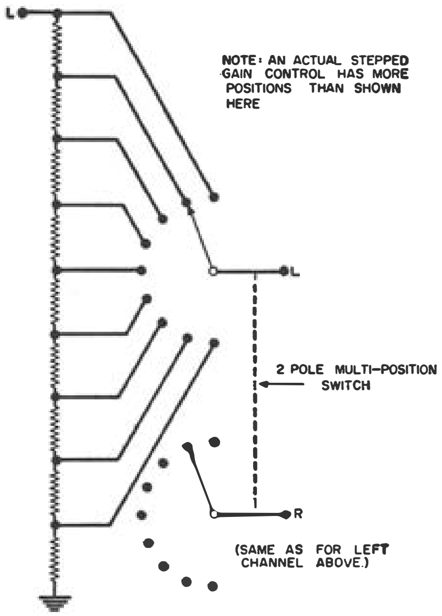
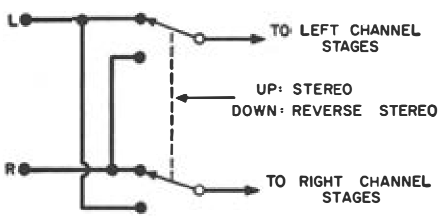


Fig. 9. Stepped master gain control.

Fig. 10. Reverse-stereo switching.



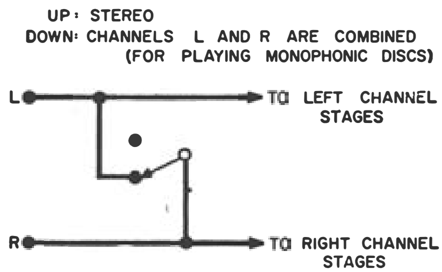


Fig. 11. Switch for combining channels.

Fig. 12. Quasi-stereo switching setup.

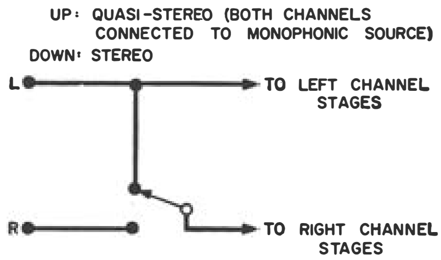
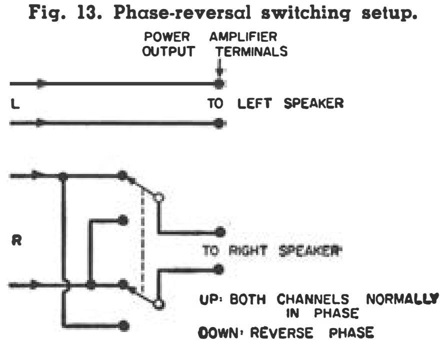


Fig. 13. Phase-reversal switching setup.



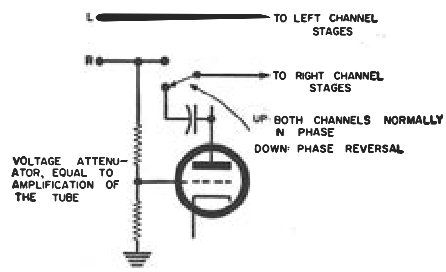


Fig. 14. Electronic phase reversal.

Fig. 15. Arrangement for blend control.

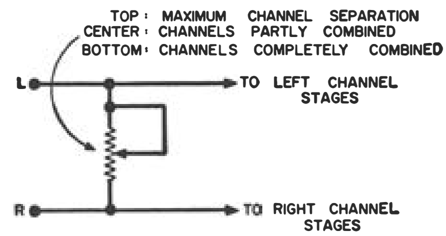
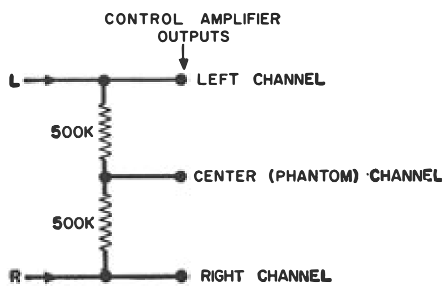


Fig. 16. Phantom channel output circuit.



ardization with respect to left-right orientation of FM-AM stereocasts. Also, amateur stereo tape recordists will probably make mistakes. (For a tape traveling from left to right, the upper track should contain the left channel signal.) Finally, the experimentally inclined stereophile will undoubtedly be curious as to the effects of reversing channels; conceivably, for some types of program material or for certain listener locations, the results may be beneficial.

2. *Balance Switching:* This permits one of the signals, say the left, to be fed either to the left or right channel, as shown in Fig. 8. Alternate switching of one signal to either channel facilitates balancing the sound levels of the two speaker systems. If the stereo amplifier incorporates reverse switching, separate balance switching is not strictly necessary, because reverse stereo accomplishes the same thing provided one disconnects one of the signals (say the right) from the amplifier. However, having to remove one of the signal sources may prove inconvenient, as for example when playing a stereo disc.

3. *Monophonic Disc Switching:* This combines the left and right signals and feeds the sum signal to both channels, as illustrated in Fig. 11. This is desirable when playing a monophonic disc with a stereo cartridge because the audio (lateral) signals are added in-phase, while the rumble (vertical) signals are added out-of-phase, causing substantial cancellation of vertical rumble, which is often a considerable problem when using a stereo pickup.

4. *Quasi-Stereo Switching:* This feeds the left (or right) signal to both channels, as shown in Fig. 12, permitting one to hear a monophonic source on both speaker systems, thus providing a quasi-stereo effect. While a system that provides monophonic disc switching (Item 3) in a sense also permits quasi-stereo switching, the reverse is not true. For example, some stereo amplifiers can feed the left signal to both channels but they cannot also feed the right signal to both channels. To sum up, it may be desired to feed *both* signals to *both* channels (monophonic disc switching) or it may be desired to feed *one* signal to *both* channels (quasi-stereo).

5. *Phase Reversal:* This permits the phase of one channel to be reversed 180 degrees. If the stereo amplifier is an integrated unit, containing power amplifiers, this can be accomplished by simply reversing the leads to one set of speaker terminals, as shown in Fig. 13. If the unit is solely a control amplifier, phase reversal must be accomplished electronically, as shown in Fig 14, by an extra tube stage. The unwanted amplification is offset by a voltage attenuator, either a voltage divider at the grid, as shown in Fig. 14, or a feedback circuit from plate to grid. While disc and tape program sources are careful to maintain proper phasing, nevertheless the circumstances of microphone and speaker placement may cause phase relationships such that a change of 180 degrees on one channel will provide better listening. Improper speaker phasing, whatever the reason, will produce poor spatial orientation, especially of sounds supposed to emanate from an area about half-way between the two speaker systems.

Blend Control

The "hole-in-the-center" effect frequently encountered in stereo can be mitigated to some extent by feeding some of the right signal to the left channel and some of the left signal to the right channel. The blend control, Fig. 15, found in some stereo amplifiers (and sometimes called a dimension control) permits the listener to combine the two signals to any desired extent.

Another method of eliminating the "hole-in-the-center" is to combine the left and right signals and feed their sum to a center speaker. The combined signals are called a phantom channel, after telephone practice. Fig. 16 shows the basic method employed by some stereo control amplifiers for deriving the phantom channel.

Stereo Glossary

By HERMAN BURSTEIN

To understand stereo you must know the language. Here are the most important terms and their meanings.

TOPICAL LISTING OF TERMS COVERED IN GLOSSARY

FORMS OF AUDIO REPRODUCTION
1. Monaural
2. Monophonic
3. Binaural
4. Stereophonic
5. Pseudo-Stereo
6. Coded Stereo

SOUND CHANNELS
7. Channel
8. Phantom Channel

STEREO OVER THE AIR
9. Stereocasting
10. Simulcasting
11. Multicasting
12. Multiplexing

STEREO ON PHONO DISCS
13. Dual-Groove Record
14. Single-Groove Record
15. Carrier-Frequency Stereo Disc
16. CBS Stereo Disc
17. Minter Stereo Disc
18. Vertical-Lateral Stereo Disc (Sugden; London)
19. Westrex Stereo Disc (45-45; Vector)

STEREO ON TAPE
20. In-Line Head
21. Stacked Head
22. Staggered Heads

FEATURES OF STEREO AMPLIFIERS
23. Balance Control
24. Focus Control
25. Master Gain Control
26. Tracking
27. Phase Reversal Switch
28. Speaker Reversal Switch

STEREO MICROPHONE TECHNIQUES
29. Left-Right Recording (Classical)
30. Listening Angle Principle
31. Longitudinal Recording
32. Mid-Side Recording (M-S)
33. Stereosonic Recording

DIFFERENCE FREQUENCY PRINCIPLE
34. Difference Frequency
35. Sum Frequency

STEREO PROBLEMS
36. Hole-in-the-Center Effect
37. Dummy Speaker
38. Matching
39. Cross-Talk

THE advent of stereo has added new terms to the audio vocabulary and given special meaning or emphasis to old terms. The following glossary seeks to explain most of the stereo terms that, at the time of writing, appear of significance to the audiophile. Common usage rather than semantics underlies these definitions. Unless otherwise stated, it should be understood that reference is to two-channel stereo, the only kind generally available to the public at the present time. Terms are not defined generally but only insofar as they apply to stereo usage.

Since an alphabetical arrangement of stereo terms would make for helter-skelter reading, they have been arranged according to topic instead. In order that the reader may quickly locate a term in which he is interested, a "Topical Listing" is included, left. Any term in the listing can be located in the glossary by number.

Forms of Audio Reproduction

1. Monaural: Audio information on one sound channel. Usually, although not necessarily, associated with one speaker system.

2. Monophonic: Same as monaural. Term coined as the counterpart of stereophonic.

3. Binaural: Audio information on two sound channels, intended for reproduction by earphones, one for each channel. Usually recorded by two microphones about six inches apart, with an intervening partition, thereby simulating the manner in which the human ears intercept sound.

4. Stereophonic: Audio information on two (or more) sound channels, intended for reproduction by an equivalent number of speaker systems. Recorded by various techniques. Stereophonic recording is still in a highly experimental stage and therefore it is not yet possible to identify stereophonic reproduction with particular types or placement of microphones.

5. Pseudo-Stereo: Devices and techniques for obtaining from a single channel source some of the qualities associated with stereo. Simplest method is to feed the single-channel source to two speaker systems spaced several feet (or more) apart. One method, commercially known as "Xophonic" (made by *Radio Craftsmen*), acoustically delays all the frequencies by about 1/20th second by passing them through a long tube then feeding them to a second speaker. Other devices operate electronically, achieving a time delay which varies with frequency; this tends to have the effect of spatially distributing the various orchestral instruments. Also see Coded Stereo, Item 6.

6. Coded Stereo: Known in one form as "Perspecta Sound," employed in theaters. Consists of single channel audio accompanied by a sub-sonic code signal that controls the volume of sound fed to speakers at the left, center, and right. Thus if drums are to appear on the left, the coded sub-sonic signal causes the system to supply relatively more power to the speaker on the left when the drums are playing.

Sound Channels

7. Channel: A signal pathway for conveying audio information during recording, transmission, or reproduction; also refers to the audio information so conveyed. Where more than one channel is employed, they differ from each other in at least one, but not necessarily all, of the following respects: frequency content, relative amplitude of various frequencies, phase, arrival time, reverberation. Typical stereo employs two channels, one intended for a speaker at the left front of the listener and the other for a speaker at the right front. By operating both channels at once, the total sound is conveyed to the listener, including a sense of direction. The left channel and speaker are identified by the number 1 or letter A. The right channel and speaker are identified as 2 or B. In recording, a third sound channel is often employed, designed to pick up the sound in the center. However, for conventional stereo, which employs only two channels in the final program source (radio, disc, or tape), the signal of the center channel is mixed with the program material of both the left and right channels.

8. Phantom Channel: Some stereo reproducing systems employ a speaker in the center as well as at the left and right, although the program source contains only two channels. The signals in the left and right channels are electrically combined to form a phantom channel, which is fed to the center speaker.

Stereo Over the Air

9. Stereocasting: Broadcasting two sound channels for stereo reproduction. There are now three techniques, listed according to present extent of usage: simulcasting, multicasting, and multiplexing. These are described in Items 10, 11, and 12.

10. Simulcasting: Broadcasting a stereo program by means of an AM and FM station (usually jointly owned). Requires an AM and an FM tuner.

11. Multicasting: Broadcasting a stereo program by means of two FM stations. Requires two FM tuners.

12. Multiplexing: Broadcasting a stereo program by means of a single FM station. One sound channel is broadcast in the conventional manner. The other channel frequency modulates a subcarrier of either 67 kc. or 42 kc. The conventional FM tuner detects the first channel and the subcarrier. A multiplex adapter, which in the future may possibly be incorporated on the same chassis as a conventional FM tuner, extracts the second channel from the subcarrier.

Stereo on Phono Discs

13. Dual-Groove Record: Employs an outer set of grooves for one channel and an inner set for the other channel. Requires two cartridges side-by-side for playback. Cartridge alignment must be precise so that the stylus of each will enter the proper groove. This method is no longer in commercial use.

14. Single-Groove Record: Employs a single set of grooves for stereo material and requires only one cartridge with one stylus for stereo playback. There are two basic methods: those employing both vertical and lateral modulation of the groove; and those employing only lateral modulation (in the manner of a monaural record) together with a modulated carrier frequency. The vertical-lateral techniques are embodied in the Vertical-Lateral Stereo Disc, the *CBS* Stereo Disc, and the *Westrex* Stereo Disc (although in this latter method the cutting angles are shifted 45 degrees). The lateral techniques include the Carrier Frequency Stereo Disc and *Minter* Stereo Disc.

15. Carrier-Frequency Stereo Disc: One channel is cut laterally in the usual manner. The second channel is employed to frequency modulate a supersonic carrier frequency, which is also cut laterally. The playback cartridge delivers the signal for one channel plus the carrier frequency containing the other channel. The carrier must then be demodulated to furnish the second channel.

16. CBS Stereo Disc: Developed by the *Columbia Broadcasting System,* this disc is cut vertically and laterally. But it may be played back on a 45-45 cartridge, which has two elements, one responding to stylus motion at an angle of 45 degrees to one side of vertical and the other element responding to stylus motion at a 45-degree angle to the other side of vertical (See discussion of *Westrex* Stereo Disc). The sum frequency of the two sound channels, A + B, is cut laterally. The difference frequency, A—B, is cut vertically. This is done at a much reduced level. In playback, the 45-45 cartridge acts as a matrixing device, so that one of its elements delivers essentially an A signal while the other delivers essentially a B signal.

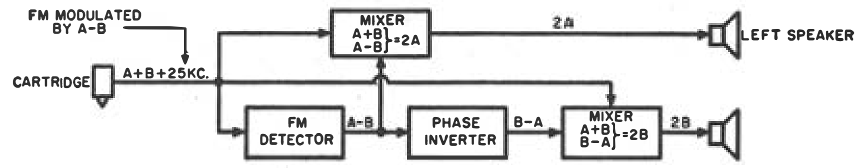


Fig. 1. Simplified block diagram of electronics used in playback of Minter disc.

17. Minter Stereo Disc: The sum of the two channels, namely A + B, is cut laterally. The difference between the two channels, A—B, is obtained electrically and employed to frequency modulate a 25 kc. carrier, which is also cut laterally. In playback, a monaural type pickup (responding only to lateral stylus motion) with relatively flat response to 30 kc. is used. While the cartridge immediately delivers the A + B information, special electronic equipment is used to obtain the A — B information and to mix it with the A + B signal so as to derive separate A and B signals. The technique for doing this is illustrated in Fig. 1. The A — B signal is extracted by an FM detector and combined with the A + B signal to produce A information; then the A—B signal is phase inverted to form B—A, which is combined with the A + B signal to produce B information.

18. Vertical-Lateral Stereo Disc: Identified with the names of *Sugden* and of *London Record Co.* The groove is cut vertically for one channel and laterally for the other. The playback cartridge contains two elements, one responding to vertical stylus motion and the other to lateral stylus motion.

19. Westrex Stereo Disc: Also known as the 45-45 Disc or Vector Disc. The record groove is in the form of a "V" and each wall of the "V" is at 45 degrees to vertical. The left wall is recorded so that it contains channel A information (for the left speaker) and the other wall contains channel B information. The signal for the left speaker causes the stylus to move at a 45-degree angle to vertical, namely from bottom left to top right; that is, the left side is cut in a manner that causes the stylus to move slantwise along the *right* wall. The right channel causes the stylus to move from bottom right to top left, along the left wall. A combination of signals from both channels causes the stylus to move in some intermediate position. The cartridge employed for playback —the so-called 45-45 cartridge—contains two elements, one responding to stylus motion at an angle of 45 degrees to right of vertical and the other responding to stylus motion at an angle of 45 degrees to left of vertical.

Stereo on Tape

20. In-Line Head: Consists of two tape heads in a single casing, one mounted directly above the other so that their gaps are in exact vertical alignment. If the stereo tape runs from left to right, the upper head reproduces (or records) the left channel and the lower head the right channel. The in-line playback head is suitable only for a recorded stereo tape with one channel directly above the other. It is not suitable for a staggered tape (see Item 22).

21. Stacked Head: Same as In-Line Head.

22. Staggered Heads: Separate heads, spaced about 1¼" apart, for playing (or recording) the upper and lower halves (tracks) of a stereo tape. If the tape runs from left to right, the head on the right is for the right channel and operates on the lower track of the tape. Staggered heads are suitable only for the recorded tapes with tracks staggered in corresponding fashion. Staggered heads and staggered tapes are virtually obsolete.

Features of Stereo Amplifiers

23. Balance Control: Device on a stereo amplifier to vary the volume of each speaker system relative to the other, at the same time maintaining their combined volume virtually the same. As one speaker increases and the other decreases in volume, sound appears to shift from left to center to right or *vice versa.*

24. Focus Control: Same as Balance Control.

25. Master Gain Control: Device on a stereo amplifier to simultaneously control gain of both channels.

26. Tracking: A function of the master gain control in a stereo amplifier, namely to maintain the same *difference* in volume between the two channels at all settings of the control. This difference, which may be zero (representing equal volume from each speaker), depends upon the setting of the balance control. Good tracking exists if the master gain control at any setting does not cause the difference between channels to change by more than 1 or 2 db.

27. Phase Reversal Switch: Device on a stereo amplifier, or in a speaker system, for shifting phase by 180° on one channel. This can mean merely interchanging the two leads to one of the speaker systems. If stereo speakers are improperly phased relative to each other, sound may appear to come from the center instead of having wide spatial distribution. Improper phasing can also lead to partial cancellation at some frequencies due to one speaker diaphragm moving in while the other is moving out.

28. Speaker Reversal Switch: Device on a stereo amplifier for connecting the left channel to the right speaker and *vice versa.* May be a means of

correcting for improper left-right orientation in the program source.

Stereo Microphone Techniques

29. Left-Right Recording: Also known as classical stereo recording. Microphones are placed at left and at right on a line parallel to the sound source. Usually the microphones are placed 6 to 20 feet apart; sometimes more in order to enhance the effect of spatial distribution. For binaural reproduction (through earphones), the microphones are usually placed about six inches apart, with an intervening object to simulate the human head. Sometimes, although infrequently, the latter technique is employed for stereophonic purposes. At frequencies where the stereophonic effect is most pronounced, namely above 1000 cycles, there are substantial phase differences in the sounds reaching each of the two closely spaced microphones. Hence, even though the speakers used in reproduction are several feet apart, there can be some kind of stereophonic effect resulting from pickup by microphones only six inches apart. When microphones are spaced a substantial number of feet apart, often a center microphone is also employed. At some stage in the recording process, the sound of the center channel is added to left and right channels.

30. Listening Angle Principle: A principle sometimes employed in Left-Right Recording. The microphones at the left and right, as shown in Fig. 2, are spaced so that they are on the angle formed between a listener in a favorable seat at the original performance and approximately the extreme ends of the music source. It is intended that the same angle should be formed between the listener and his two speaker systems, more or less. Hence the microphones and the speakers, through a common angle, in effect attempt to put the listener in the "favorable seat" he might have occupied at the original performance.

31. Longitudinal Recording: Microphones are spaced along a line at right angles to the music source (*i.e.,* from front to back). The result is a time delay between channels, as well as differences in amount of reverberation; typically, the microphone closer to the source picks up more direct and less reverberated sound.

32. Mid-Side Recording: Referred to as M-S Stereophony. See Fig. 4. Employs one cardioid microphone and one cosine (figure-8 polar pattern) microphone very close together. The cardioid is oriented to pick up all the audio information, which may be called A + B + C, with A representing the left, B the right, and C the center. The cosine microphone is placed so that its figure-8 reception pattern is parallel to the sound source, thereby picking up more of the sound on the left (A) and on the right (B) than in the center. The A sound picked up by the cosine microphone is 180° out-of-phase with the B sound inasmuch as the microphone has but one pressure-sensitive element, which obviously cannot move two ways at once. Hence the sound picked up by the cosine microphone may be called A—B. Fig. 4 shows how the signals of the two microphones are combined. The A—B signal plus the A + B + C signal produces a 2A + C signal. The A—B signal is then combined out-of-phase (thus becoming B—A) with the A + B + C signal, producing a 2B + C signal. In sum, one channel contains information principally from the left, and the other contains information principally from the right; each also contains some center information.

33. Stereosonic Recording: Similar to Mid-Side Recording, it employs two microphones very close together and relies largely upon intensity differences in the signal picked up by each one. As shown in Fig. 3, each microphone has a figure-8 pattern, oriented 45° to the sound source. One microphone picks up sound essentially from the left, while the other picks up sound essentially from the right. If desired, a combined output signal can be obtained from the two microphones in a manner similar to that shown in Fig. 4 for Mid-Side Recording.

Difference Frequency Principle

34. Difference Frequency: Signal representing, in essence, the difference between the left and right sound channels, namely A—B. Usually obtained by simple electronic means (one signal is phase inverted 180° and then added to the other signal); also achieved by microphone techniques (see Mid-Side Recording, Item 32). Use of a difference frequency in recording, radio transmission, etc, is an application of information theory, enabling audio information to occupy a smaller portion of the chosen medium; that is, the difference between two like signals is less than either signal alone. In the case of stereo discs, use of the difference frequency for vertical modulation of the groove results in a smaller vertical cut than if either of the original channels governs the amount of vertical modulation.

35. Sum Frequency: Sum of the left and right sound channels. See discussion of Difference Frequency, Item 34.

Stereo Problems

36. Hole-in-the-Center Effect: Apparent absence or insufficiency of sound in the region between the left and right speakers. This effect may occur if the recording microphones and/or stereo speakers are placed too far apart.

37. Dummy Speaker: A psychological device to overcome the Hole-in-the-Center Effect, consisting of a speaker system, or merely a speaker enclosure, placed between the left and right speakers. Although no signal is fed to the central speaker, nevertheless for some persons the visual presence of the middle speaker helps create the aural illusion of sound from the center.

38. Matching: Refers to use of speakers with identical or very similar frequency response characteristics. Speakers with unlike response—peaks and valleys at different parts of the audio spectrum—may cause the apparent source of sound to wander between left and right.

39. Cross-Talk: Undesired reproduction on one channel of audio information intended for the other channel. Occurs to a slight extent in in-line heads, where magnetic coupling causes the upper head to pick up from the lower head some of the signal which the latter has picked up from the lower track of the tape; the lower head of course picks up the upper track signal in similar fashion.

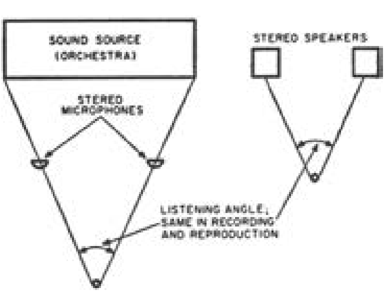


Fig. 2. The listening angle principle.

Fig. 3. Miking in stereosonic recording.

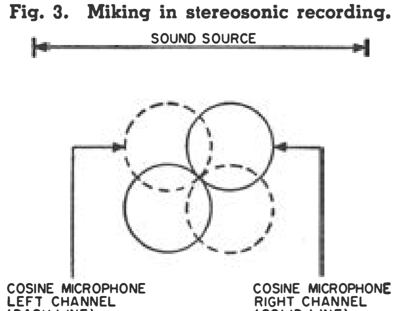
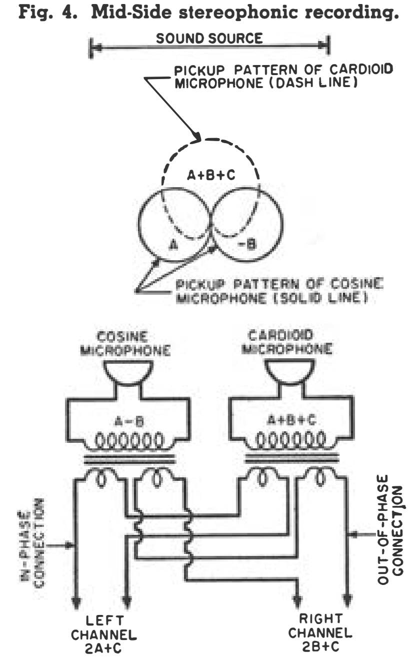
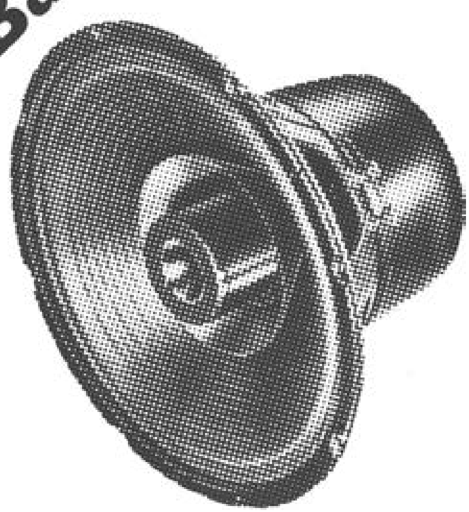
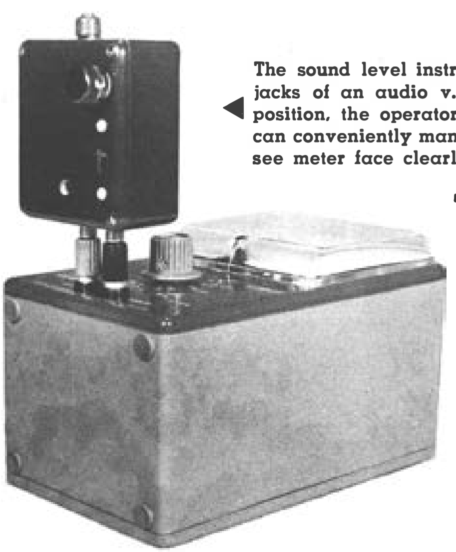


Fig. 4. Mid-Side stereophonic recording.





Stereo Sound Balance Meter



◀ The sound level instrument is plugged directly into the input jacks of an audio v.t.v.m. With the meter in this horizontal position, the operator, when standing behind the instrument, can conveniently manipulate the controls on the meter and can see meter face clearly. Sound unit front faces sound source.

By **HAROLD REED**

THE device to be described in this article is designed as an accessory to an audio vacuum-tube voltmeter or a.c.-d.c. voltohmmeter—the combination converting the voltmeter into a sound-level indicator which provides visual indication of the acoustic output level from any loudspeaker. It is a valuable addition to the test equipment line-up of both audiophiles and technicians engaged in setting up stereophonic systems. The device was designed with low cost, compactness, and ease of operation as basic criteria.

One of the most important considerations in a stereo setup is the balancing of the sound level from the two loudspeakers. The most favorable speaker locations in any particular room depend on the room acoustics and the sound level desired, as well as on the personal preferences of the listener. Trying to obtain the correct sound balance by ear does not always give the best results as one is never quite sure that proper balance has been obtained.

Circuit Description

The schematic diagram of this v.t.-v.m. accessory is given in Fig. 1. A small 2½" PM speaker is used as a microphone to pick up sound output from the stereo speakers. Its 3.2-ohm voice coil works into transformer T_1, a 3.2:1000-ohm unit. An impedance of 1000 ohms is about right for the input circuit of the first transistor stage, V_1. The audio signal is fed to the base of V_1 through capacitor C_1. Proper bias and stabilization of the first transistor stage is provided by resistors R_1 and R_2. Coupling to the base of the second stage, V_2, is through capacitor C_2 and volume control R_4. Bias and stability conditions are taken care of by the resistance of the volume control and R_5. Audio output for the indicating meter is obtained through capacitor C_3 which is connected to the output terminals.

Power for the circuit is supplied by a miniature 15-volt battery. The minute power output from the small speaker-microphone is, therefore, amplified to a suitable level to provide a satisfactory reading on an a.c. vacuum-tube voltmeter.

Construction

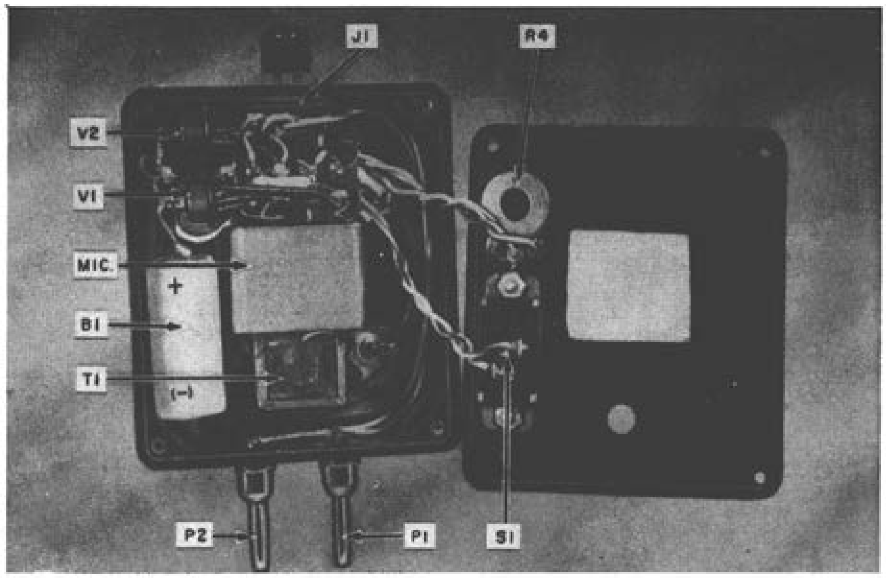
The little device is assembled in a miniature 3" x 2⅜" x 1⅜" plastic speaker baffle. This baffle was designed for use with the speaker selected for this application.

A small piece of perforated Bakelite board, equipped with solder terminals, is used as a tie point for the various components. This board may be made from the Bakelite strips and the flea clips that are readily available. The terminal board was fitted with two small home-made metal brackets which were soldered to the speaker framework. The transistors, resistors, and capacitors were soldered directly to the terminals as shown in one of the photographs. The battery is connected to these same terminals with #16 insulated leads. The back of the speaker case fits snugly against the battery and this, together with the heavy wire leads, keeps the battery firmly in place. The tiny matching transformer is also soldered directly to the metal framework of the speaker.

The small, slide-type power switch and miniature gain-control potentiometer are mounted on the back plate of the speaker baffle. This is convenient, since the device is normally used with the operator standing at the rear of the unit as will be described later. Flexible wire leads are used for connection to the gain control and power switch.

The rear of the speaker is quite close to the back of the baffle and since the case is not designed for speaker mounting screws, a small piece of sponge rubber, about 1/8" thick, was cemented to the back plate of the baffle and when screwed into place this holds the speaker firmly in position. If desired, the speaker face may be cemented to the front of the baffle. The transistors, which probably won't have to be replaced, and the battery may be removed without detaching the speaker from the case.

Output connections are taken to two banana plugs attached to the bottom of the speaker case. These plugs are spaced 3/4" apart so the device may be plugged directly into the input terminals of an audio v.t.v.m., as shown in the photograph. The baffle comes



Inside view of the sound level meter with all important parts clearly labeled.

equipped with a built-in jack and miniature plug. This jack is connected in parallel with the banana plugs. By means of the miniature plug, the output may be fed to other type meters or to a tube or transistor power amplifier for other applications.

Substitute Parts

The constructor may build this device using the parts given in the parts list; however, some readers may wish to substitute components from their "junk boxes." For instance, CK722 transistors can be employed in place of the 2N107's without any circuit changes or 2N180 transistors may be used. The latter will provide higher gain than the other two types.

Likewise, speakers with other voice-coil impedances can be substituted by using a suitable matching transformer. Different transformers may be connected in the input circuit—the parts list suggesting only two of the available units. An input impedance of 1000 ohms is about right for feeding into the base of the first transistor stage. The author used the less expensive 3.2: 500-ohm transformer, type TR-95, with satisfactory results.

Further, any small speaker may be built into any suitable metal box or cabinet in place of the plastic baffle suggested. Other output connectors than the ones given may be substituted and coupling capacitors can be any value from 1 to 6 μfd.

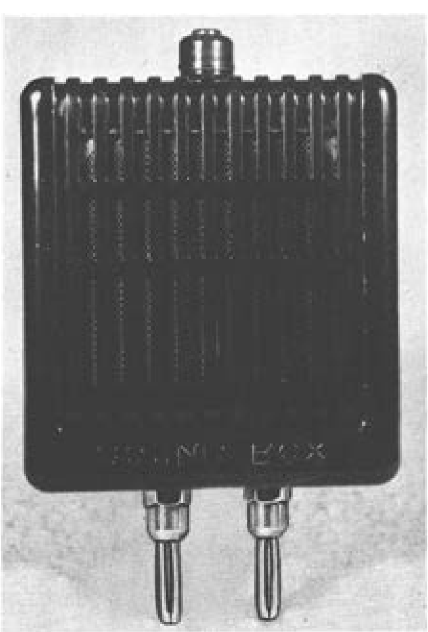
The unit shown was designed with both satisfactory results and low cost as objectives. The parts for the unit, as described, cost about $7.00.

Operating Data

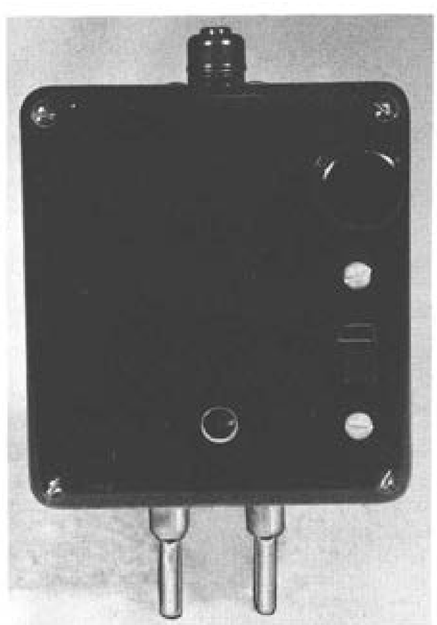
It has been recommended that the speakers of a stereophonic system be placed about 8 feet apart. Also, it has been specified that the listening position be from 12 to 15 feet from the speakers and midway between them. Using these recommendations, the sound balance meter should be set up as shown in Fig. 2. It can be placed on a small table at the listening position. Due to space limitations it may not always be possible to attain the "ideal" embodied in the recommendations, but irrespective of the physical arrangement, the important point concerning balance measurement is that the measuring device be placed at the listening location and oriented so that it is facing toward the mid-position between the two speakers.

Have the power switch, S_1, off and the volume control, R_1, turned all the way down. Set the v.t.v.m. switch to the 0.3-volt range. Feed the output of an audio oscillator, at 1000 cycles, to channel 1 of the stereo system. Adjust the volume control of this channel for moderate sound output from the channel 1 speaker. Stand behind the sound

Front view of the sound level instrument. The banana plugs are at the bottom and the miniature plug and jack, supplied with the speaker case, are at the top of the plug-in unit.



Rear view of the sound level instrument. Note that the miniature volume control potentiometer and the slide-type power switch are attached to the right of the back plate of unit.



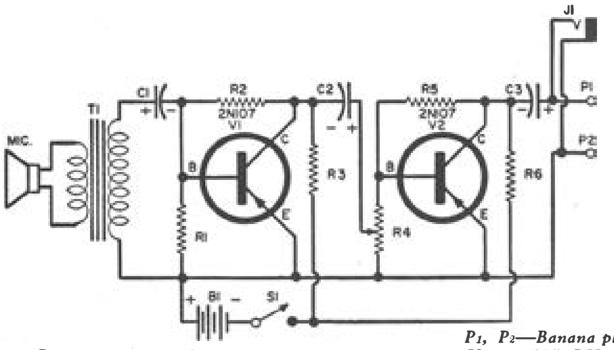


Fig. 1. Complete schematic diagram and parts listing for the sound level meter. Note that V_1 and V_2 may be 2N107's, CK722's, or 2N180's. Parts are not critical and substitutes may be used.

R_1—12,000 ohm, 1/2 w. res.
R_2—120,000 ohm, 1/2 w. res.
R_3—24,000 ohm, 1/2 w. res.
R_4—10,000 ohm miniature pot
R_5—150,000 ohm, 1/2 w. res.
R_6—7500 ohm, 1/2 w. res.
C_1—2 μfd., 6 v. elec. capacitor
C_2, C_3—2 μfd., 15 v. elec. capacitor
S_1—S.p.s.t. slide switch
J_1—2-conductor, open-circuit jack (included with speaker baffle)
P_1, P_2—Banana plug
Mic.—2 1/2" PM speaker, 3.2 ohm v.c. (used as microphone)
B_1—15-volt battery (Eveready #411 or Burgess U10)
T_1—Transistor trans. 3.2 ohms to 1000 ohms (Argonne AR-138 or Lafayette TR-95, see text)
Speaker Baffle—3" x 2 3/8" x 1 3/8" speaker baffle (Lafayette MS-315)
V_1, V_2—"p-n-p" transistor (G-E 2N107. See text for other types)

level meter and throw the power switch on and adjust its volume control to obtain a suitable indication on the audio v.t.v.m. The v.t.v.m. may be switched to the 0.1-volt range if necessary, depending, of course, on the sound level from the speaker. Mid-scale on the meter is a good reference point. Note this meter reading.

Now, feed the oscillator output to channel 2 of the stereo system and again stand behind the sound level meter. Note the meter reading obtained from the channel 2 speaker. If it is higher than the reading noted when feeding the signal through channel 1, decrease the gain control of channel 2. Likewise, if the meter reading is lower, increase the gain control of channel 2. The purpose of this adjustment is to obtain the same reading from both speakers. The volume control of the sound-level unit should not be changed once it is adjusted for a reference reading with the 1000-cycle signal being transmitted through the first channel. When comparing the levels from each channel, as indicated on the v.t.v.m., it is important that these readings be observed from the same position directly behind the sound-level device, otherwise erroneous readings may result due to different sound-wave patterns reflected from the body of the operator when standing in different positions. This is a precaution to be taken when using any type of acoustic sound measuring instrument.

Balance adjustment may also be accomplished by using an audio tone test record on the record player of the system or with a test signal from a magnetic tape instead of the audio oscillator suggested previously.

Another factor to be considered for favorable stereophonic listening is speaker phasing, that is, the speaker cones should move forward and backward in unison. A sound image occurs midway between the two speakers when the sound level and audio frequency reproduced by them are identical and the signals are in-phase. Out-of-phase speakers in the same room cause a certain amount of sound-wave cancellation and results in thin sounding, instead of full bass frequencies.

Feeding the audio oscillator to the input of both channels of a stereo system and adjusting the channel gain controls for equal sound levels from each speaker, the effect of out-of-phase speakers was detected by the author on the sound balance meter. When the speakers were connected from out-of-phase to in-phase condition, the sound level increased 6 db as read on the db scale of the audio voltmeter. The best audio frequency for making this test is 400 cycles since it provides the greatest change in the meter reading. These results will vary with different installations. The important point in any case is to connect the speakers for maximum indication on the meter.

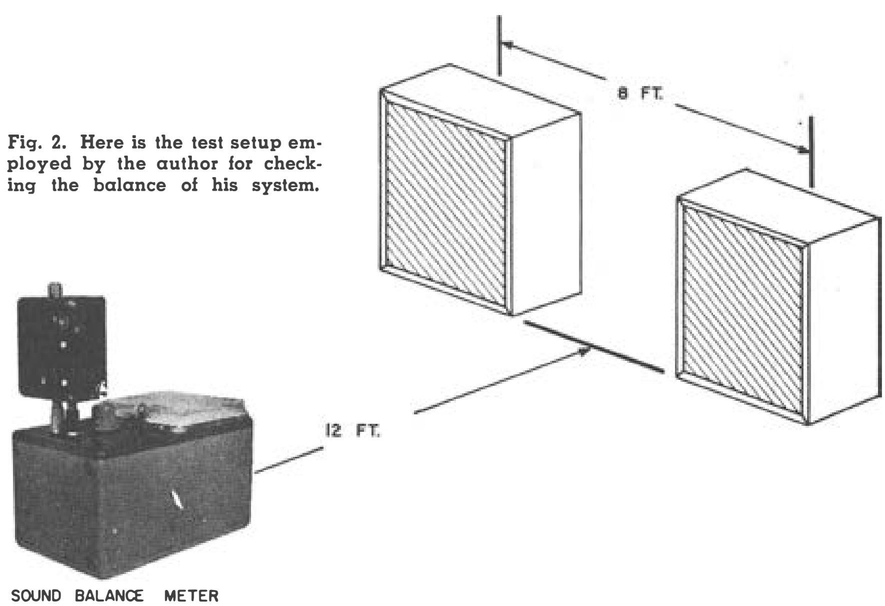
Another use for this sound-level device is in comparing the efficiency of identical speakers. The speakers, in turn, are placed at a distance, say 5 feet, from the sound-level instrument and the relative sound outputs noted on the meter. In making this test, each speaker must be placed successively in exactly the same position, the same frequency (400 or 1000 cycles) must be used, and the same signal level maintained as read across each voice coil with a v.t.v.m. It is preferable to make this measurement with the speakers and test instrument in an open area as far as possible from surrounding objects in order to minimize the effects of reflected sound waves. Also, it is necessary to stand in the same spot behind the sound-level meter, as previously explained.

The reader will now realize that this little instrument can be used for many relative sound-level measurements, both in comparison tests and in ascertaining when a sound level has increased or decreased when altering certain environmental conditions. Besides its use as a sound-level device, it may be employed as a general purpose two-stage amplifier and can be worked into a vacuum-tube or transistor power amplifier.

It was mentioned previously that an a.c.-d.c. voltohmmeter could be used as the indicating meter. An *RCA* "Volt-Ohmyst" was tried and worked satisfactorily. Of course, it is not as sensitive as the audio v.t.v.m. but with the switch in the 1.5-volt position and a higher sound-level output from the stereo loudspeaker, satisfactory balance measurements were made.

The life of the battery will be long since the total current drain is just 1.7 milliamperes.

Fig. 2. Here is the test setup employed by the author for checking the balance of his system.



Section II

Amplifiers, Preamplifiers and Record Players

SINGLE PUSH-PULL STAGE FOR BOTH STEREO CHANNELS

By **NORMAN H. CROWHURST**

Simple simplex-type circuit for stereo does away with two output tubes and one output transformer.

IF STEREO can be recorded in a single groove, why cannot it be amplified by a single amplifier? As with so many questions, this one has two possible answers: it can't be done; and the people who do it! In this case the latter are *CBS Laboratories,* as reported in a paper before the Audio Engineering Society, jointly authored by B. B. Bauer, W. S. Bachman, J. Hollywood and G. Maerkle.

The question, "How does it work?", which this article aims to answer, can likewise be asked with different attitudes: the man who said it can't be done has objections, and doesn't think it can work *properly;* while the person who is unprejudiced just wants to know, in simple terms, the principles involved, as well as "Does it do a job as good as two separate amplifiers, of the same, or lower cost, or with the same total output?".

In an ordinary push-pull amplifier, all the tubes and other components of the push-pull part are in duplicate, and handle audio exactly the same, except that one "pushes" when the other "pulls" For good push-pull operation, both "halves" of the amplifier carry identical waveforms, except that one swings up when the other swings down. Usually great care is exercised to ensure the two halves are balanced so the waveforms really are identical.

But actually a push-pull amplifier is two separate amplifiers, the only tie together being at the input, or phase inverter, and the output, a push-pull transformer. Failure to maintain the ideal balance would not cause any trouble until the two are recombined at the output. So what is to stop each side of the "push-pull" stage being used for one channel of stereo, instead of going to all that trouble to get exact identity for just one output? And when you look at it, the principle is quite simple (although one can always say that when someone else has already done it!). In fact it's as simple as making each half carry the modulation from one side of the record groove in a 45-45 record (Fig. 1).

By now it is well known that, when the two channels work together, as they do for a center-located sound, the groove moves from side to side without any change in depth (Fig. 1C). When only one channel carries program, due to a sound originating from one extreme side, only one wall of the groove is modulated (Fig. 1A or 1B). And when the two work in opposition, the groove goes directly up and down (Fig. 1D).

This last condition does not normally happen at lower frequencies, because it would represent a sound "off-stage". But it can and does happen at higher frequencies, because the time difference can then amount to several wavelengths.

From Fig. 1 it will be seen that the center-located sound gives the normal push-pull waveform combination, while the out-of-phase condition gives "push-push". Stereo program would be mono if it only contained the push-pull combination, but on the other hand, very little of it reaches the completely push-push condition of simple up-and-down. Most of it lies somewhere between these extremes.

(Most common stereo cartridges are phased in such a way that lateral motion produces in-phase signals. By simply reversing the connections to one of the pickup elements, the phase conditions shown in the figure are obtained. With 4-terminal cartridges this is simply a matter of transposing 2 leads; with 3-terminal cartridges the manufacturer must provide the required phasing. See Ques. 2.—Editor)

If one pick-up output were fed into

Fig. 1. The relationship between various types of grooves on a 45-45 disc and the outputs from the stereo cartridge discussed in the article. Although coils are shown, ceramic elements would produce the same results. (A) and (B) show sound in one channel only, while both channels have equal signals in (C) and (D). In (C) the cut is completely lateral, in (D) it is vertical.

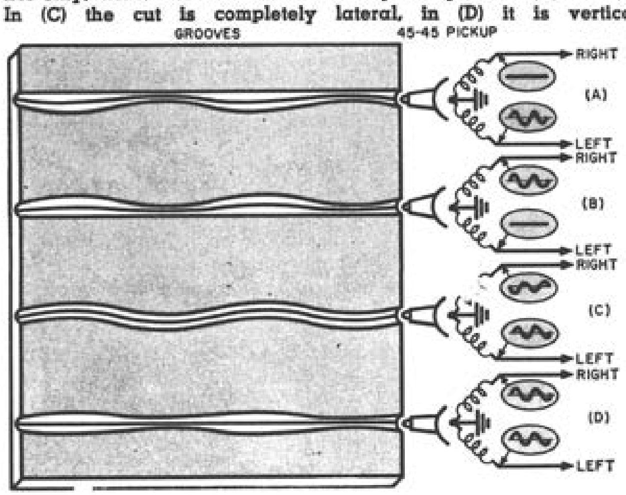


Fig. 2. The double-matrixing transformers operate push-pull and "push-push", or single-ended, to produce these waveforms.

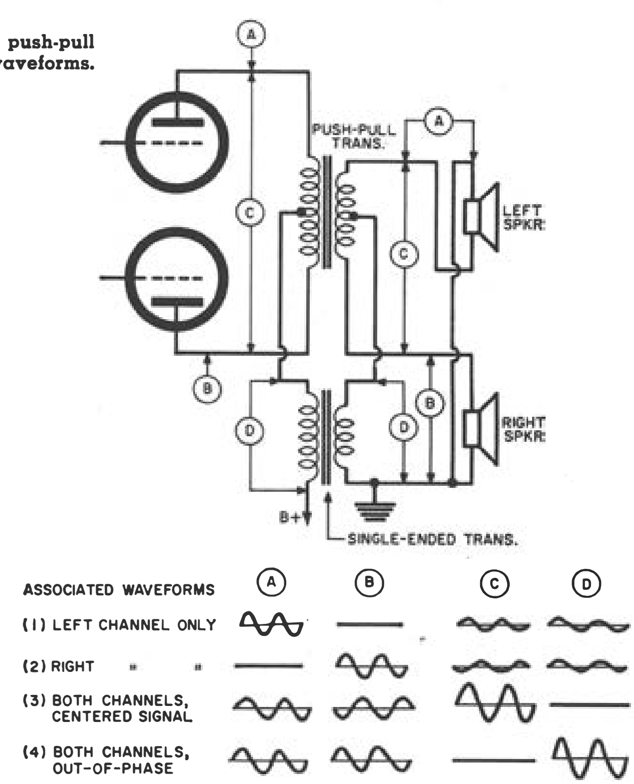


Fig. 3

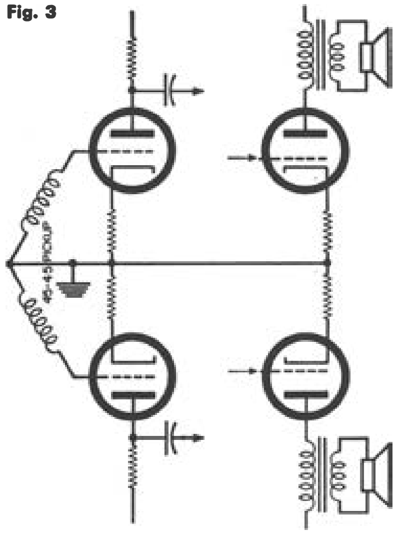


Fig. 3. With single-ended output transformers, a full-length, push-pull amplifier could be used as two separate single-ended amplifiers as described in the text.

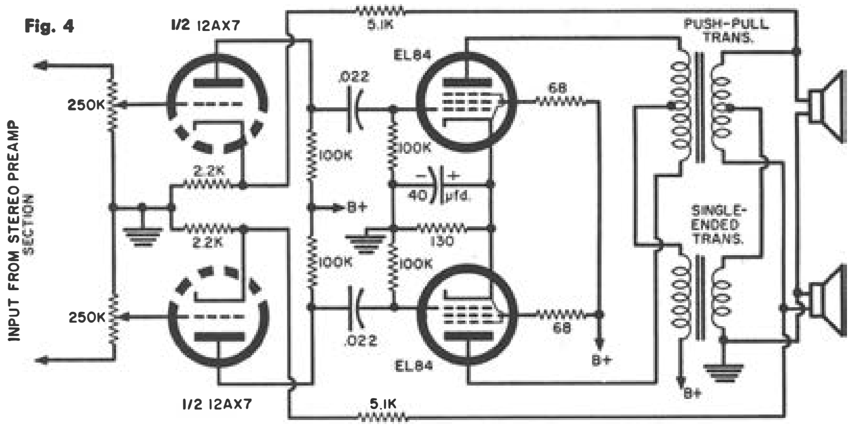


Fig. 4. Schematic diagram of driver and output stages, showing use of feedback.

Fig. 6. Class B operation is a condition not completely realizable in practice: it depends on "curves" with straight lines and sudden corners (A); practical tubes have bends (B).

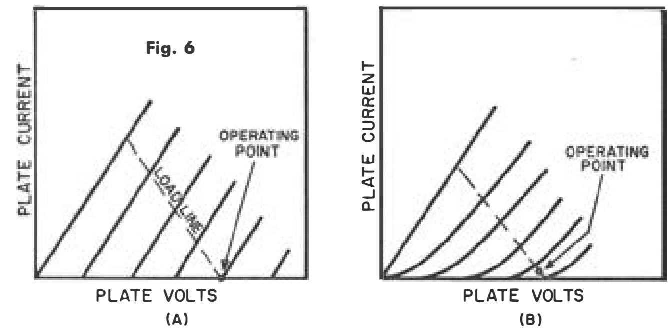


Fig. 5

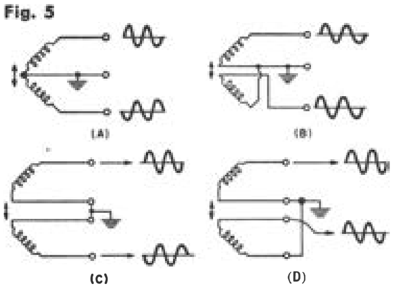


Fig. 5. The two possible ways a 3-terminal cartridge can be connected internally (A) for CBS push-pull system and (B) for "regular" separate system. A 4-terminal cartridge can be connected (C) for CBS system and (D) for "regular" stereo system.

each side of the so-called push-pull stage, and each side had a separate output transformer feeding its own loudspeaker, we should have a couple of separate amplifiers working from a common power supply, of the quality normally expected using single-ended output stages (Fig. 3). The kernel of the new development is the double matrixing (mixing) output circuit that effects an economy in output transformer requirements, and *at the same time* enables the normal advantage of push-pull output to be obtained.

Instead of using one output transformer for each channel—left and right—separate transformers handle virtually the "lateral", or push-pull and "vertical", or push-push components (Fig. 2). Remember, the out-of-phase condition never normally happens in stereo program at low frequencies, and only stands a random chance of happening at higher frequencies.

So the transformer that carries the two plate currents in parallel does not need a good bass response. Thus the normal objection to a single-ended output—loss of bass—is avoided in having the transformer acting single-ended. The *CBS* paper also claims an advantage in downgrading bass response to the "vertical"—a built-in vertical rumble rejection, that certainly can often be helpful.

The other transformer acts strictly push-pull, and thus is able to have all the qualities of a push-pull output transformer. Now we begin to see where the saving comes in. Only *one* high quality push-pull output transformer is needed; the other can be smaller and much cheaper. And we need only one push-pull output stage, as regards all the other components, through which to feed stereo program material.

Feedback is taken from the resultant output to the voice coils, back to the cathodes of the driver stage (Fig. 4). This can reduce distortion in either channel (left or right), correct frequency response, and reduce any error in the double-matrixing action of the output transformers.

That about tells the story as far as the principle is concerned. But a new idea like this will start (in fact it has started) some questions, with the idea "Does it really buy all this?". So let's take some of these questions, as a way of exploring the potentialities of this kind of amplifier.

1. You said the push-pull transformer has all the advantages of a normal push-pull output transformer. I can see that the static, or quiescent plate currents will balance and thus maintain its inductance and low frequency response; but isn't part of the function of a normal output transformer to cancel even order distortion from the amplifier? How can this happen when the amplifiers are handling different channels?

This objection would be true for separate, single-ended output transformers (Fig. 3). But with this arrangement, the push-pull transformer only handles that part of the composite program content that is strictly push-pull. The "single-ended" component is handled by the smaller transformer. There is, in almost any stereo material, a dominance of high amplitude lower frequency component almost in phase. Any distortion of these components in the amplifier *is* cancelled by the push-pull transformer in the same way as a regular push-pull output.

Then, in the final amplifier circuit, overall feedback, from the individual speaker connections, goes back into the amplifier to linearize each channel *as an entity*, regardless of its division into mono and stereo (or push-pull and push-push) components.

The major form of distortion reduced by the normal push-pull output transformer is this lower frequency component. The push-pull transformer here does it too, both as regards harmonic and IM components. Distortion higher up, which gets more complicated anyway, is taken care of by the feedback, as it also is in any normal amplifier.

2. Will not the loudspeakers, connected as at Fig. 4, reproduce the stereo out-of-phase?

This is a matter of phasing, *at both input and output*. Most three-terminal pickups are phased so that lateral motion of the stylus gives two outputs that are positive at the same time (Fig. 5). In this system, the pickup has to be connected so, for lateral motion, one output is positive at the instant the other is negative.

In the phonographs using this amplifier, the pickup is phased correctly for the purpose. Using the amplifier with other pickups is no problem when there are four terminals, so the user can phase his pickup to suit.

If you've become used to thinking the way most systems are connected is *standard,* then this system will seem non-standard to you in this respect. But it really depends which you start off with as the standard. From the viewpoint of the cutter, or pickup, basic symmetry would require the up-and-down component to be the in-phase, or *push-push element,* which agrees with this system, and makes the more generally accepted connection "non-standard".

As far as *functioning* is concerned, all that matters is that it be connected the *right* way, which only has to be done once, when setting up the system.

At the loudspeaker end, if identical units are used, the opposite end of the voice coil(s) have to be connected to the ground bus on the left and right systems.

3. One reason, or advantage of push-pull operation is improved efficiency of the output stage. This is achieved by working at, or nearer, class B operation. In view of the fact that this system uses a push-push component, can it utilize this advantage properly?

Class B is a theoretical condition, postulated on plate characteristics that make an abrupt, or discrete transition from straight lines representing constant a.c. resistance, to a cut-off condition, representing infinite resistance (Fig. 6). No tubes ever made operate just this way.

But working well "round the bend" as a single-ended tube can result in excessive curvature. And the parallel, or vertical component does pass through this amplifier as a "single-ended" operation. Let's give a little thought to what can happen in a single-ended amplifier of this type with feedback.

Assume it is biased to a point well in the curvature, and that a signal comes through that drives it "round the bend"—in fact well into cut-off. Feedback tries to offset the waveform inside the amplifier that is distorted in the opposite way (Fig. 7). But it can only work on parts of the waveform that get through. For the part beyond cut-off there is no feedback.

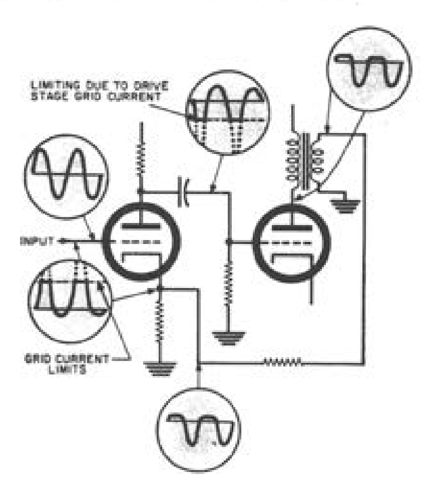
So the internal waveform becomes exaggeratedly asymmetrical the opposite way. As has been proved many times, such an asymmetrical component in a waveform is equivalent to a change in d.c. bias; in this case it will work progressively, like a "pump", until the feedback can "get to work" on the whole waveform.

In effect, the feedback will use the time constant of the coupling between drive and output stage to alter the bias just enough to allow the stage to handle the signal completely, so it can work on linearizing all of it.

But, if this were the only means, and the time constant is made long enough to represent a good bass response (which is needed for the push-pull mode), quite a bit of distortion can occur *before* the bias gets readjusted. Fortunately, however, with the circuit shown in Fig. 4, another effect can take charge meanwhile.

When the large signal first "strikes", its first positive excursion at the grid of the drive stage (Fig. 7), it won't be offset by corresponding feedback at the cathode, because the output tubes will run well into cut-off. Consequently, *from the point where the output tubes cut off,* the grid voltage here will rise sharply positive, and due to grid current, will temporarily bias this stage back by a corresponding amount. With proper choice of time constants, this will pass a similar temporary bias to the output stage—positive, so the output stage can immediately handle the whole signal.

Fig. 7. Waveforms in single-ended amplifier with feedback when the output tube is driven beyond cut-off. See text.



4. Doesn't having the tubes handle a "double" signal—push-push as well as push-pull—limit the maximum power of the amplifier as compared with normal push-pull operation, in spite of any self-adjusting action?

To tackle this question I went to the Mullard "Technical Handbook of Receiving Valves" to see what I could expect of a couple of EL84's.

In Class AB, self-biased push-pull, with 300 volts on the plates, they give 17 watts. A figure is not given for the same operating condition in parallel, but an inspired guess from figures given for a single tube operating at 250 volts suggests they would give around 13 watts under this condition.

Working in this circuit, if a pure signal is fed in, in-phase in both channels, so as to work the tubes in push-pull, they will give their rated 17 watts, into the rated resistance load. This will be shared between the left and right channel, after an appropriate loss in the output transformers.

Similarly, if one channel is reversed in phase, to be equivalent to a vertical cut, the tubes should deliver 13 watts into the same kind of load. So, when someone asks whether this mode of operation will limit the power, do they want to get 17 + 13 watts = 30 watts, for the same money they can normally get 17 watts?

Actually, under the hypothetical conditions represented in such tests, the amplifier should always be able to deliver somewhere between 13 and 17 watts, according to phase angle between channels.

But actual stereo program does not possess a single frequency with known or constant phase difference between channels. Different components will have different phase angles, at quite random distribution. But this is nothing new. The same invalidation of wattage ratings occurs with normal amplifiers.

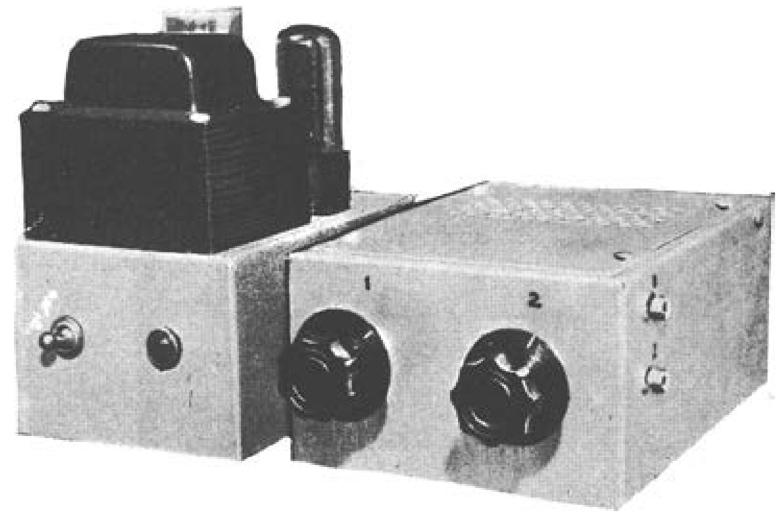
An amplifier is never called upon, in musical program, to deliver 17 watts pure sine tone into a resistance load. It's called on to deliver a multiplicity of complex tones into a loudspeaker. *If it had a resistance load,* the maximum peak power of the complex wave would be just twice the average power of the theoretical sine-wave output. That's about as nearly as we can relate the measured results to practical performance.

5. What about crosstalk in a combined amplifier like this?

The answer to this question requires qualification. It depends on what the crosstalk is. If it is pure crosstalk—left program breaking through to right, or vice versa, 10 to 12 db separation is quite adequate. But if it should happen to be distortion components of left program showing up in right, then 30 or 40 db is not good. So, when you measure crosstalk, do you have simple crosstalk, or instead do you have *cross-intermodulation?*

Pickups, as well as combined amplifiers, will need more careful scrutiny from this viewpoint. There is no reason why a combined amplifier should not have a separation of better than 25 db that is pure crosstalk, due to slight imbalances and tolerances on transformer ratios, etc.

Stereo amplifier, shown here, with matching power supply (left), is built in 5"x7"x3" chassis.



Simple STEREO Amplifier

By
ROBERT W. TIMMERMAN

Compact 3-tube amplifier operates directly from ceramic stereo cartridge, easily drives high-efficiency speakers.

THE amplifier described here was developed with the twin objectives of economy and compactness as a complete unit for stereo service. The resulting amplifier is in the low-power class, however its output is adequate for monitoring or for normal home listening with speakers of moderately high efficiency. It will operate directly from a ceramic-type stereo pickup cartridge. Standard circuits and readily obtainable parts are used. The cost of building the amplifier and power supply will be between $25 and $35, excluding labor. If a suitable power supply is available, parts for the amplifier itself will run around $15 to $20.

The amplifier proper is built within a 5" x 7" x 3" chassis, using somewhat unconventional layout and mechanical details which will be described later. The power supply, which may be located at some distance from the amplifier, is built on a 4" x 6" x 3" chassis.

Circuit

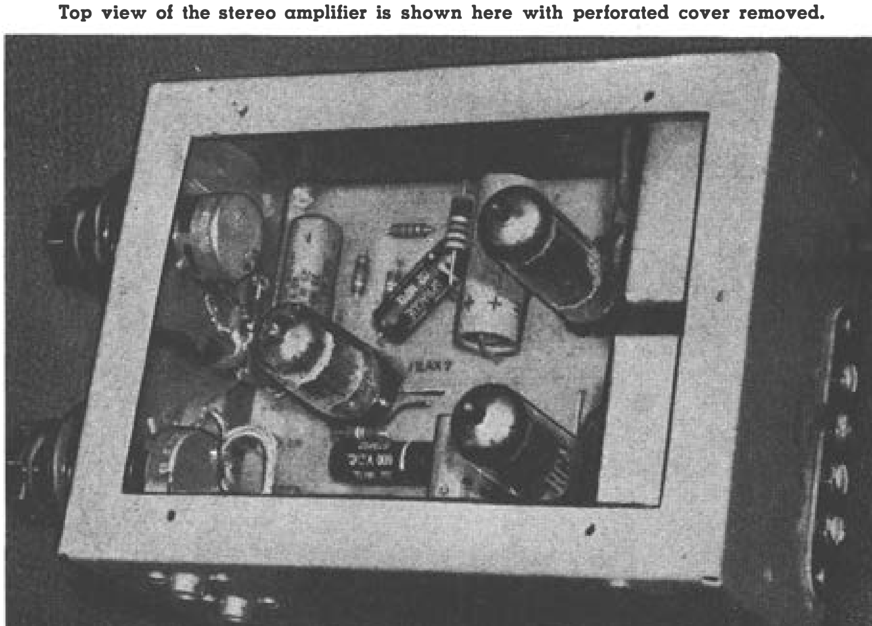
The circuit diagram of the amplifier is shown in Fig. 1. It consists of two identical, but independent, amplifiers. A twin-triode 12AX7 is the voltage-amplifying first stage for each channel. Each output stage is a type 6973 beam-power tube operating single-ended. About 8 db of voltage feedback is applied from the secondary of each output transformer to the cathode of the corresponding 12AX7 triode section. Resistors R_4 and R_3 make up the feedback voltage divider for Channel 1 while R_{11} and R_{10} perform the same function for Channel 2. Eight db is about the maximum amount of feedback that can be used with inexpensive transformers and with the limited gain available in the two amplification stages. The input signals are applied to individual volume-control potentiometers R_1 and R_8. Conventional coupling and cathode bias circuits are used. The 6973 power-amplifier tubes are operated with distributed screen (Grid No. 2) load through the use of center-tapped output transformers.

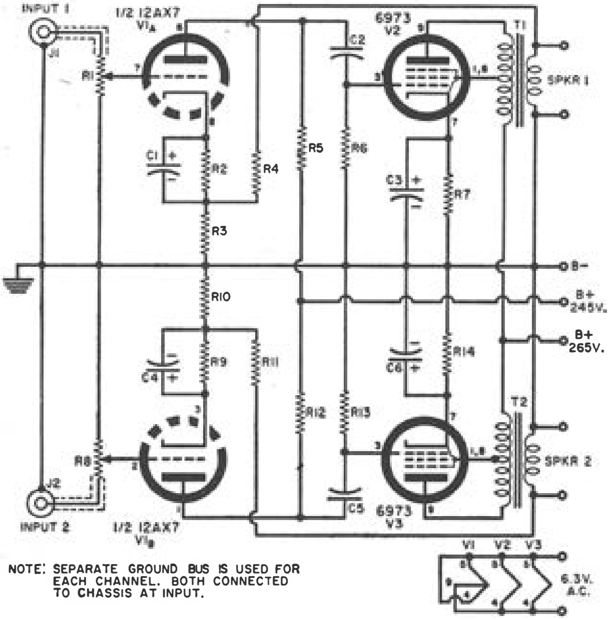
Fig. 2 is a schematic of the power-supply circuit. It is conventional with the possible exception of R_1, which is an adjustable resistor allowing precise setting of the high voltage for maximum power output with minimum distortion. Resistor R_2 and capacitor C_3 provide extra filtering for the voltage amplifier stages. A 5-wire cable connects the power supply to the amplifier.

Mechanical Features

The amplifier was planned so as to incorporate a printed (etched) circuit. While printed circuits were originated primarily for economy in mass production, this technique is also usable in home construction projects. Slightly more time is required for planning and layout, compared to conventional design, but final assembly is accomplished very quickly and a permanent, compact, and rugged unit results. Moreover, additional interest is created for the audiophile who derives pleasure from construction as well as from operation of his equipment. Information is given here to permit duplication of the author's etched-circuit amplifier. Other arrangements of parts

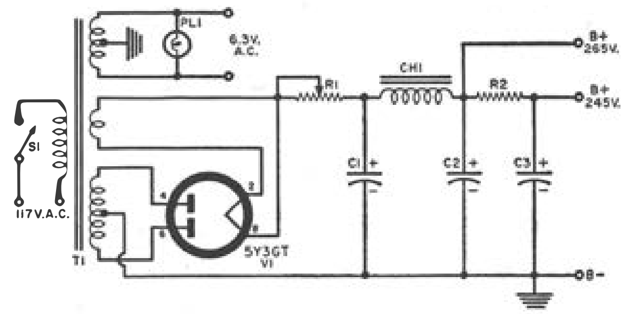
Top view of the stereo amplifier is shown here with perforated cover removed.





R_1, R_8—1 megohm audio taper pot
R_2, R_9—2200 ohm, ½ w. res. ±10%
R_3, R_{10}—100 ohm, ½ w. res. ±5%
R_4, R_{11}—1200 ohm, ½ w. res. ±5%
R_5, R_{12}—220,000 ohm, ½ w. res. ±10%
R_6, R_{13}—470,000 ohm, ½ w. res. ±10%
R_7, R_{14}—330 ohm, 1 w. res. ±10%
C_1, C_4—25 μfd., 25 v. elec. capacitor
C_2, C_5—.02 μfd., 600 v. capacitor
C_3, C_6—100 μfd., 25 v. elec. capacitor
T_1, T_2—Universal output trans., 18 w. (see text, Merit A-2904, Thordarson 22S95, Stancor A3852, or equiv.)
J_1, J_2—Shielded phono jack
V_1—12AX7 tube
V_2, V_3—6973 tube

Fig. 1. A simple circuit and inexpensive parts are used here.



R_1—1000 ohm, 10 w. wirewound adj. res.
R_2—10,000 ohm, 1 w. res.
C_1, C_2—40/40 μfd., 450 v. elec. capacitor
C_3—8 μfd., 450 v. elec. capacitor
CH_1—8 hy., 85 ma. filter choke
PL_1—6.3 v. pilot light
S_1—S.p.s.t. toggle switch
T_1—Power trans., 350-0-350 v. @ 100 ma.; 6.3 v. @ 1.5 amps; 5 v. @ 2 amps.
V_1—5Y3GT tube

Fig. 2. Power supply has provision for plate voltage adjustment.

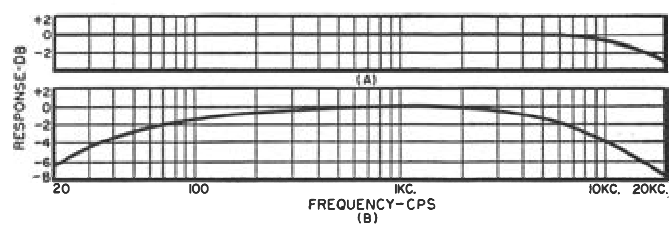


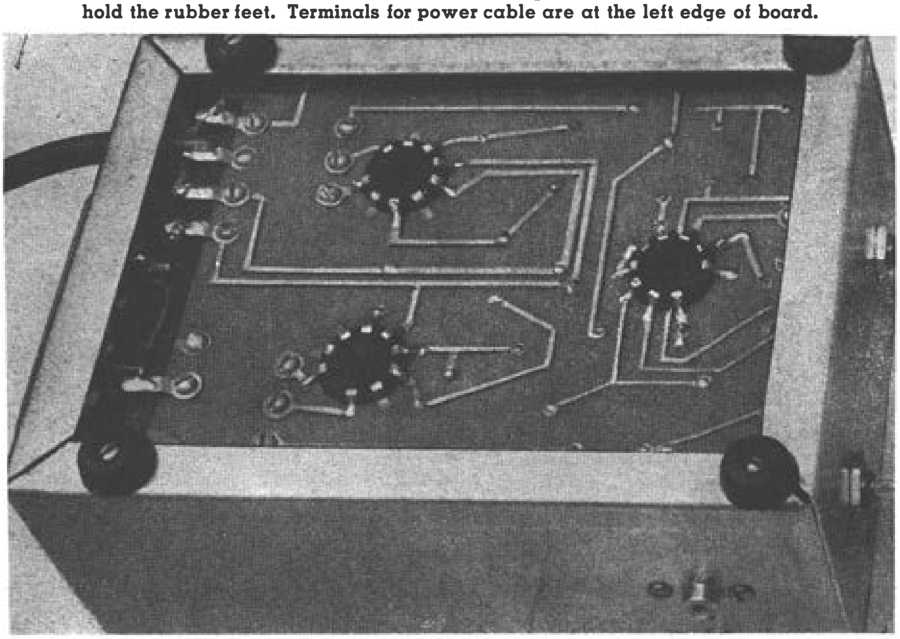
Fig. 3. (A) Response of each channel at 100 mw. output. (B) Power response for just visible distortion. 0 db = 2.3 watts.

would be satisfactory or the components can, of course, be assembled and wired in traditional style, if desired.

Components

All of the electrical parts for the amplifier and power supply are standard items, readily available from mail-order and local supply houses. The essential specifications are given in the parts lists accompanying Figs. 1 and 2 and require no comment except for the output transformers. In keeping with the economy objective of this project, relatively low-priced transformers are specified. To be assured of adequate direct-current-carrying capacity in the primaries and to go at least partway toward minimizing core saturation effects of single-ended operation, nominal 18-watt transformers are recommended, even though the maximum power output per channel is two to three watts. The "universal tube-to-voice coil" type provides a center-tapped primary and a multi-tapped secondary. Secondary taps should be chosen to give an impedance ratio of approximately 4000 ohms (full primary) to 8 ohms secondary. The primary center-tap is connected to grid No. 2 of the output tube. Only one pair of output terminals is provided for each channel, which on measurement and listening tests actually operates satisfactorily with speakers rated from 4 to 16 ohms.

Bottom view. Printed circuit board is mounted by four corner bolts which also hold the rubber feet. Terminals for power cable are at the left edge of board.



Construction

Assuming that the etched circuit technique will be used, the first step is to lay out the circuit board. With Figs. 4 and 5 and the photos as guides, locate the terminal points and conductor routes in full scale on a sheet of thin paper. Final size of the phenolic, copper-clad board is 4¾" by 5⅞". Slight adjustments may be required to accommodate the dimensions of the components to be used. After completing the layout and making a thorough double check, the conductor lines are traced onto the copper face of the circuit board with carbon paper. The writer has found it convenient at this stage to mark with a center punch each point at which a hole is to be drilled. These points include terminals of all components as well as input, output, and power-supply connections and the centers of the tube-base circles.

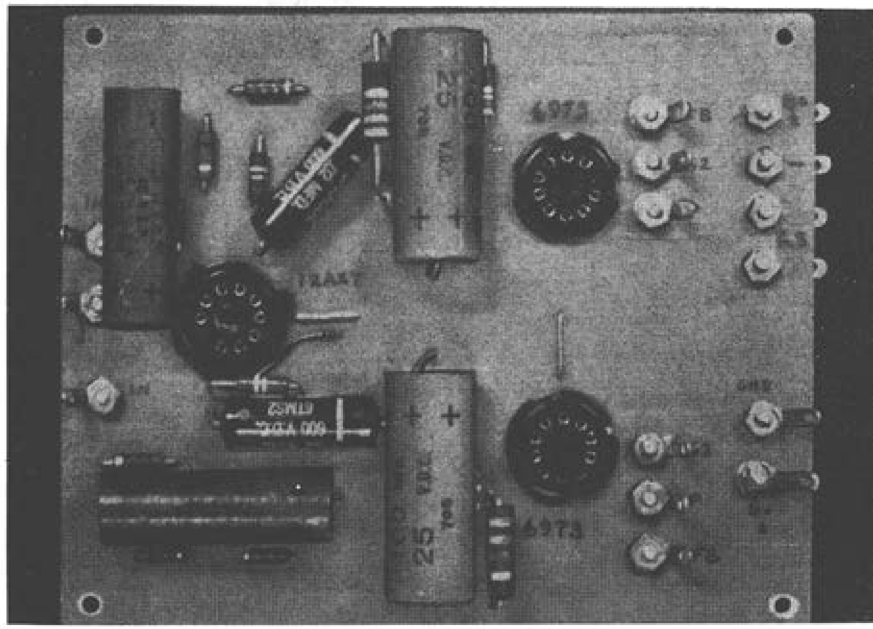
Application of the "resist" ink is next. This technique has been described by Middleton and Marshall in the August 1954 issue of *QST* and the reader is referred to this article or other sources for details. *Union Ink Company* Type C-992 resist was used by the author. Tabs for the tube connections should be matched to the sockets. Snap-in sockets, such as *Eby* PC-9, are recommended. Terminals for external connections should be made about $\frac{3}{16}''$ in diameter to provide firm connections to soldering lugs fastened by No. 4 bolts. Etching away of the unwanted copper areas is then carried out with warm 30% ferric chloride solution. The resist is finally washed off with solvent, leaving the copper conductor pattern exposed and intact.

The circuit board is prepared for the mounting of the parts by cutting the tube socket holes, filing them carefully to size, and then drilling small holes for the pigtail terminals. A No. 60 drill is the correct size for most components. One-eighth-inch holes are drilled at the external connection points. Pigtails of the components are then bent to fit the required mounting centers and the parts are mounted in their proper places on the top side of the board. The tinned pigtails should extend about $\frac{1}{32}''$ out from the circuit side. Each pigtail is soldered to its terminal, using minimum amounts of solder and heat. The tube sockets are inserted in proper orientation and each terminal is soldered in place. Small soldering lugs are attached by 4-40 bolts, lock washers, and nuts at the input, output, and power-supply terminals. This type of connection is preferable to soldering of external wires directly to the etched circuit conductors, as strain on the wire could readily pull loose the copper foil.

After completion of the circuit board, attention may be given to preparation of the cabinet. A standard 5" x 7" x 3" aluminum chassis is the starting point. Holes are drilled for mounting the input jacks, volume controls, power cable, and the four-point output terminal strip. Locations of these items are not critical and may be judged from a study of the photographs. A rectangular opening is cut in the top of the chassis, leaving a $\frac{5}{8}''$ rim all around the entire top edge.

The circuit board is mounted by four 6-32 bolts, spaced $\frac{1}{4}''$ above the bottom with short pieces of tubing. The four rubber feet are also held by these board mounting screws. The feet are important since they elevate the amplifier above the operating surface to provide necessary ventilation. The circuit board is located near the front of the cabinet, leaving about $\frac{3}{4}''$ between the back edge of the board and the back inside of the cabinet. The latter space is for the power cable and speaker terminal connections.

Shielded input leads are provided from the jacks to the volume controls. The controls are oriented to permit the shortest possible connections to the board input terminals.



Top view of the printed circuit board after all the components have been mounted.

Frequency Response: (100 mw.)	Flat 20-8000 cps; down 3 db @ 20,000 cps
Power Output: (low distortion)	2.3 w. mid-range; down 3 db @ 45 and 8000 cps
Input:	0.45 v. across 1 megohm for full output
Inverse Feedback:	8 db
Output Matching:	8 ohms nominal; feeds 4 to 16 ohms satisfactorily
Hum and Noise:	−73 db
Crosstalk:	−51 db
Damping Factor:	3.2

Table 1. Performance data for each of the channels in the stereo power amplifier.

Fig. 4. Bottom view layout of the $4\frac{3}{4}$x$5\frac{7}{8}$-inch printed circuit board. Black lines show locations of copper conductors. Terminal points are shown by enlarged spots. Note that this drawing is not shown full size but is included for guidance only.

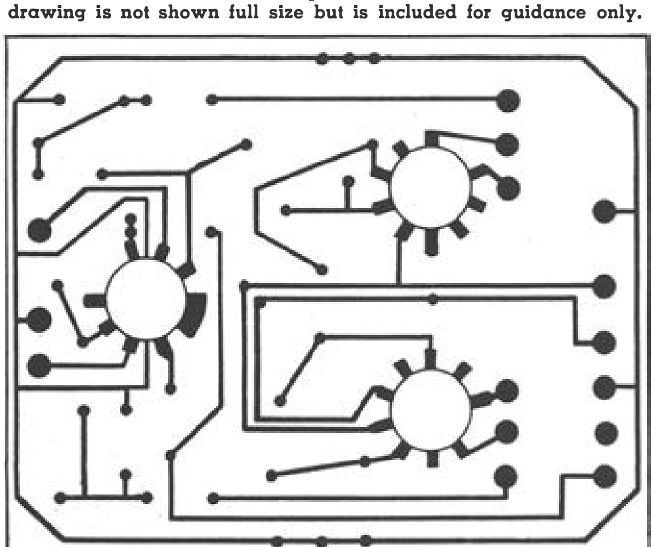
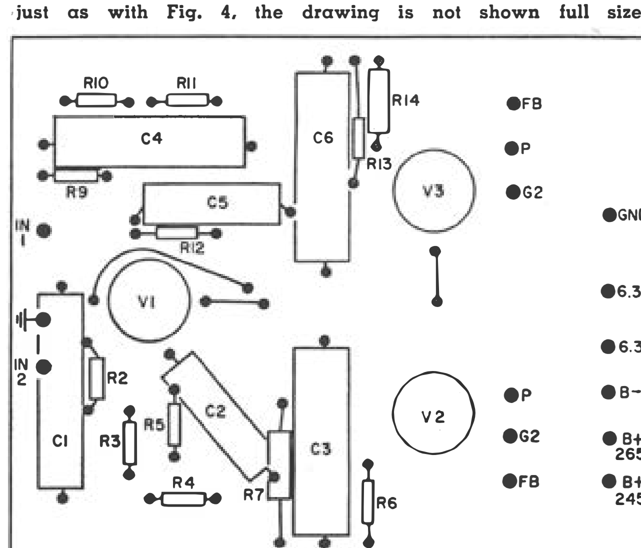


Fig. 5. Locations of components on the circuit board. This is a bottom "x-ray" view; the parts are actually mounted on the top side, opposite that of the etched circuit. In this case, just as with Fig. 4, the drawing is not shown full size.



The output transformers are installed last. Part of one mounting ear on each unit must be cut away to mount in the limited space available. The cut ear is located up and is held against the side of the cabinet by a 6-32 bolt and nut. The lower ear is wedged into the space between the circuit board and side of the cabinet. Connections must be made to the transformers by flexible wires before the transformers are slipped into their final positions.

Output (voice-coil) connections must be made with the correct polarity to provide negative feedback. Proper connections for *Merit* A-2904 transformers are as follows: Primary: brown, "B+"; red, Grid No. 2; blue, plate. Secondary: No. 1, hot output and feedback; No. 6, ground output. Other makes of transformers may require different connections.

Assembly of the amplifier is completed by attachment of aluminum cover plates to top and bottom. Both covers must have ventilation holes. The amplifier becomes quite hot in operation and if it is to be mounted in a warm or congested location, more holes than shown in the photos should be drilled and, in addition, the sides of the cabinet should be perforated.

The power supply is assembled on a 4" x 6" x 3" chassis, using conventional mounting and wiring procedures. Power requirements of the amplifier are 265 volts d.c. at 100 ma. and 6.3 volts a.c. at 1.2 amps.

Testing and Operation

For the first trial under power, R_1 (Fig. 2) should be set to a point near its maximum resistance. The power and line cables are then connected and the line power switch is turned on. After checking to see that all tubes are lighted and no components are seriously overheating, several voltage measurements are made. The "B+" 265-volt terminal should now measure about 200 volts above ground (chassis). If the reading is much lower than this, turn the power off immediately and look for a possible short circuit. A high voltage at this point indicates a possible open or high-resistance circuit.

When everything appears to be normal, R_1 is adjusted until voltage at the "B+265 v." terminal reaches that value. Adjustment of this voltage is important in securing minimum distortion at maximum power output. Corresponding correct voltages at various tube pins in Fig. 1 are as follows (all positive with respect to chassis): V_1: Pins 1 and 6, 130 v.; Pins 2 and 7, 1.5 v. V_2 and V_3: Pins 1 and 8, 255 v.; Pin 9, 245 v.; Pin 7, 14 v.

The amplifier is now complete and ready for a final listening test. The pickup cartridge is connected by two shielded lines terminating in pin plugs. The speaker circuits have a common, grounded terminal, thus a 3-wire cable may be used between the amplifier and the speakers.

Reminiscing about Stereo

By **PETER L. JENSEN**

Chairman of the Board, Jensen Industries, Inc.

TO MOST hi-fi enthusiasts, stereo is a relatively new phenomenon — first brought to the public on tape and now through the medium of records. But stereo was not a sudden, overnight invention; it is the culmination of what might be called the sound epoch, the era of continual, relentless research to reproduce sounds with the finest fidelity. Stereo represents the furthest advance in that effort.

Looking back over the years in retrospect, today's stereo seems eons of time away from the first crude instruments we used 50 years ago to transmit sound —the first crystal sets. But even then, stereo already was being spawned in those days.

Whenever the history of stereophonic sound is written, due credit should go to the oldtime sports announcer named "Foghorn" Murphy. If it hadn't been for Murphy and his fuzzy reading of baseball scores at a San Francisco ball park, the first loudspeaker might not have been perfected, and stereo started on its way.

This little-known chapter in the story of sound goes back to one afternoon in 1910. Another young engineer, Edwin L. Pridham, and I had set up a small laboratory on the outskirts of Napa, California. We had been trying to develop an improved telephone receiver and instead had stumbled on a device for reproducing sounds in volume. We called it a dynamic speaker.

But we soon found that no one was interested in this new sound reception device. The big New York companies told us our equipment was too bulky. Discouraged, the two of us were sitting in our shop one day, staring at the box of coils and wires that no one seemed to have any commercial use for.

A visiting friend, just back from a ball game, broke the quiet. "Why don't you make something so we can hear that marble-mouthed announcer, 'Foghorn' Murphy, at the ball park?" he suggested. "Today I couldn't hear half the lineup."

Why not? Up to now everyone had concentrated on *communicating* sound. No one had tried to *amplify* the human voice. We quickly connected a large gooseneck horn to our receiver. Using a heavy duty microphone and special wiring, we aligned equipment, then attached the wire to a battery.

We hoped this crude apparatus would be just loud enough to carry across the room. Instead, it boomed out in a deafening roar!

Though baseball fans did not beat a path to our door, the new invention found a wider audience with the coming of World War I. On Christmas Eve, 1915, Pridham and I set up the device in San Francisco for the convenience of 75,000 persons gathered to hear Christmas carols. It was the first time such a huge throng could hear all the music without ear-straining.

In September, 1919, the ailing President Woodrow Wilson, standing inside a glass cage with microphones above his head, used our loudspeakers to address 50,000 persons in San Diego on behalf of the League of Nations.

That same year Pridham and I had our first true tryout of stereophonic sound. A San Francisco nightclub with the unlikely name of the "Hoo Hoo House" asked us to set up a sound system of mikes and speakers. The club had a unique problem; two dance floors on different floors of the building and only enough money for one band.

We solved the problem by setting up five mikes (one for each band instrument) and connecting them through five amplifiers to separate speakers on the floor above. The speakers were set up to correspond to the relative position of the instrument whose sounds they carried.

But the musician's union stopped the "Hoo Hoo's" two-for-the-price-of-one plan before the week was out! And thus, stereophonic sound died out almost as soon as it was born.

The loudspeaker went on its own way, however, finding applications its inventors never dreamed of. Pridham and I thought the most likely use for the new device would be in a public address system, which we promptly worked out.

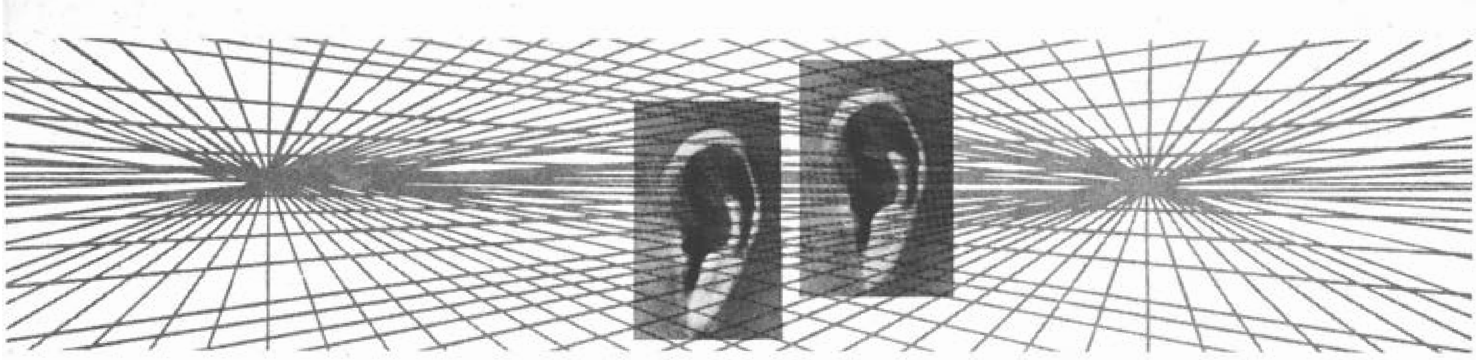
Instead, it was radio which gave the speaker its biggest boost. Before long, people were putting their headphones aside and listening directly to radios with built-in loudspeakers.

In 1925, the two of us dissolved our partnership; Pridham stayed in California to become a radio executive (later vice-president of *Magnavox Corp.*) and I moved to Chicago to open a new laboratory for designing loudspeakers.

It was not until World War II that I renewed my acquaintance with stereo —this time as it affected phonograph needles and cartridges, rather than speakers.

The future of stereo now seems assured. By 1962 the changeover to stereo should be virtually complete, with monophonic records going the way of 78's today.

If this seems an overly confident prediction, I am chastened by the thought of another forecast made back in 1907. A friend and I were witnessing the birth of wireless telephony, from a crude crystal set put together in a small Danish laboratory. We sat up all night discussing its possibilities—but our dreams were not big enough.



Low-Cost Stereo System

By R. J. MEAGHER

For less than $100, including speakers, you can enjoy stereo using this home-built dual 10-watt amplifier.

STEREOPHONIC sound can now be enjoyed without lavish outlays for equipment, as this article will prove. The stereophonic sound system to be described can easily be built by anyone who has ever made a radio or audio amplifier.

The audiophile who considers any speaker costing less than $100 inferior may not appreciate this system since the amplifier and speakers together in this setup cost less than this sum.

The author had been enjoying long-playing records using an old changer and a good fidelity amplifier unit. Then the new stereo records became available and the problem of how to take advantage of this sound "bonus" without spending a small fortune cropped up. After looking at various units and reading many articles on the subject, the author designed this particular system with two thoughts in mind. The first criterion was good stereo sound rather than a system having fancy specifications and the second was to keep costs at a minimum by using parts on hand where possible. Both objectives were met.

The Pickup

The stereo cartridge selected by the author was the *Columbia* CD compatible stereo cartridge, Model SC-1. It was installed in the tone arm of the old changer with a second shielded cable (supplied with the cartridge) added for stereo. The arm was first balanced to have zero weight since the cartridge weight provides the proper tracking pressure. This was done by adjusting the spring load, but may be accomplished with lead weights on the rear of the arm. A pressure gauge can be used to verify the recommended stylus pressure of 5 to 7 grams.

The Amplifiers

The dual-amplifier was then built using the circuit of Fig. 3. One power supply feeds both amplifiers, and uses an old TV power transformer. Such a transformer is easily obtained and pro-

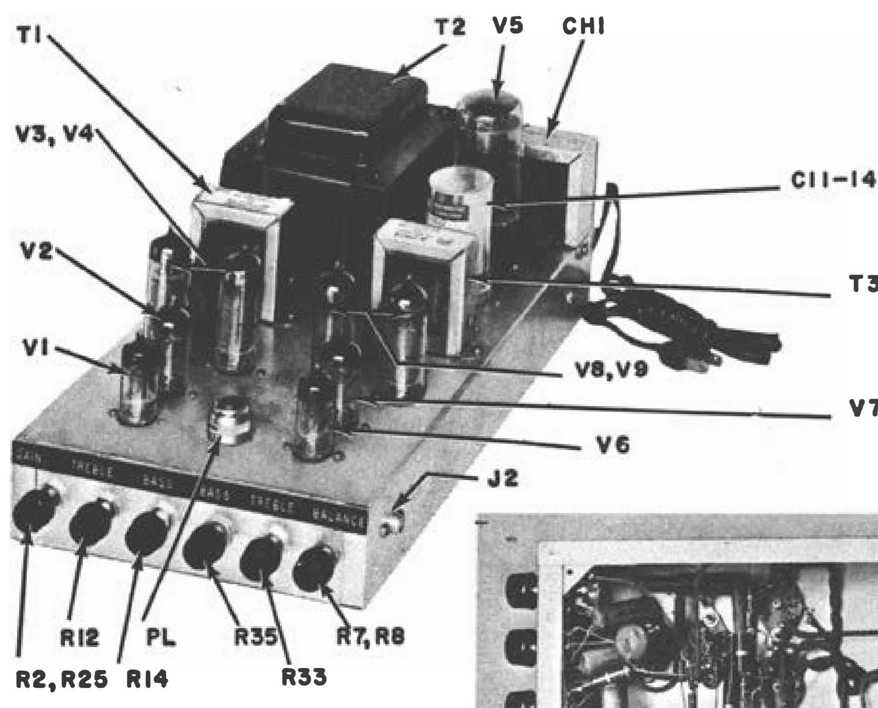


Fig. 1. An over-all view of the dual amplifier is shown in this illustration. Common power supply circuits for both channels of amplification are located at the very back portion of the chassis.

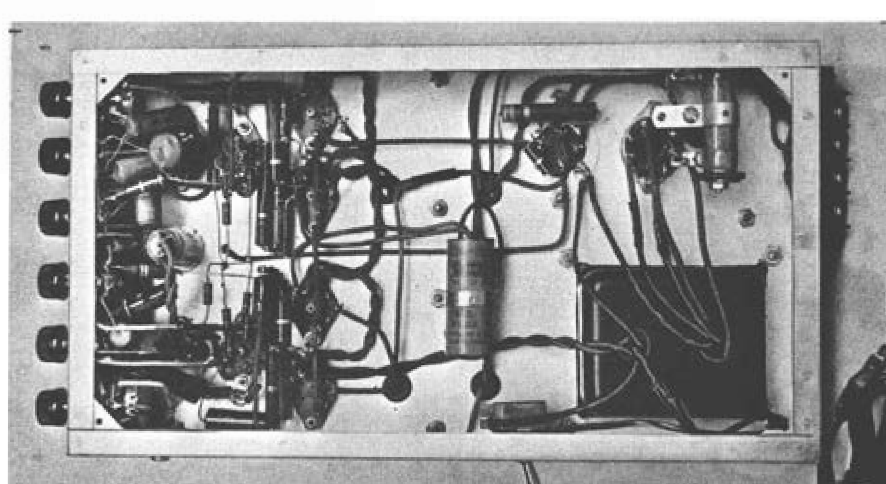


Fig. 2. Under-chassis view of the amplifier is shown here. Input terminals are on both sides of the chassis near the front, while the output terminals are on the rear panel just behind the tapped voltage-adjusting resistor R_{23}.

vides high current with good regulation. The amplifiers are identical. The 7025 (the low-noise version of the 12AX7) was chosen for its low inherent noise and hum level and the 6BQ5 for its high gain. The first stages (6C4's) are included to take care of possible low-level inputs, but since a high-output cartridge was used (the *Columbia* SC-1 is rated at 0.4 volt) sufficient gain is derived in the 7025 stage to drive the 6BQ5's. Thus, with this type of cartridge, a further cost saving can be effected by eliminating the 6C4 stage of each amplifier. No shielding was found necessary due to the short leads from the two-channel, separated layout as shown in Fig. 2. A hum balancing potentiometer was not needed because of the fortuitous choice of tubes and layout. The heater leads to the tubes should be twisted all the way and the heater ground should be made at the 6C4 end. If hum level should prove objectionable, an aluminum mesh cover can be used on the bottom of the chassis. Oscillation or motorboating may occur in either amplifier and, if so, the blue and brown leads of the output transformer involved should be reversed.

All resistors and capacitors should be chosen for small physical size since space is at a premium in the front end. All potentiometers are small ½-watt units. Considerable saving was effected by using *Merit* #2904 output transformers. They are rated at 18 watts and exhibit very satisfactory response in this circuit (run within 10-watt rating).

The purpose of the 200-ohm, 20-watt resistor between the 5U4GB and filter choke is to adjust plate voltage to within 6BQ5 ratings. They operate at about 300 volts. This will vary with different power transformers so that, in some cases, a larger resistor may be needed. R_1, R_{24}, R_2, R_{25} and C_1, C_{15} provide equalization for the SC-1 cartridge. If a different cartridge is used, these values should be changed to conform to the manufacturer's suggestions.

The positions of the line switch, input jacks, and pilot light (the latter is not shown in the schematic) were chosen only for convenience in the author's built-in cabinet and may be relocated for each individual case, taking care to keep the leads from the jacks to the tubes short and the 117-volt a.c. leads away from the high-gain inputs.

Little further need be said about the amplifier circuits, since they are straightforward. Figs. 1 and 2 show the parts layout. Except for keeping leads short to avoid the necessity for shielding, the parts layout is not critical. Be sure to place the power transformer so that its windings are at right angles to the output transformers to prevent induced 60-cycle hum, since they are close to one another.

The Controls

Referring to the circuit diagram (Fig. 3) and the front-view photograph (Fig. 1), there is a single master gain control for both channels. This control is R_2, R_{25}, a dual potentiometer, shown

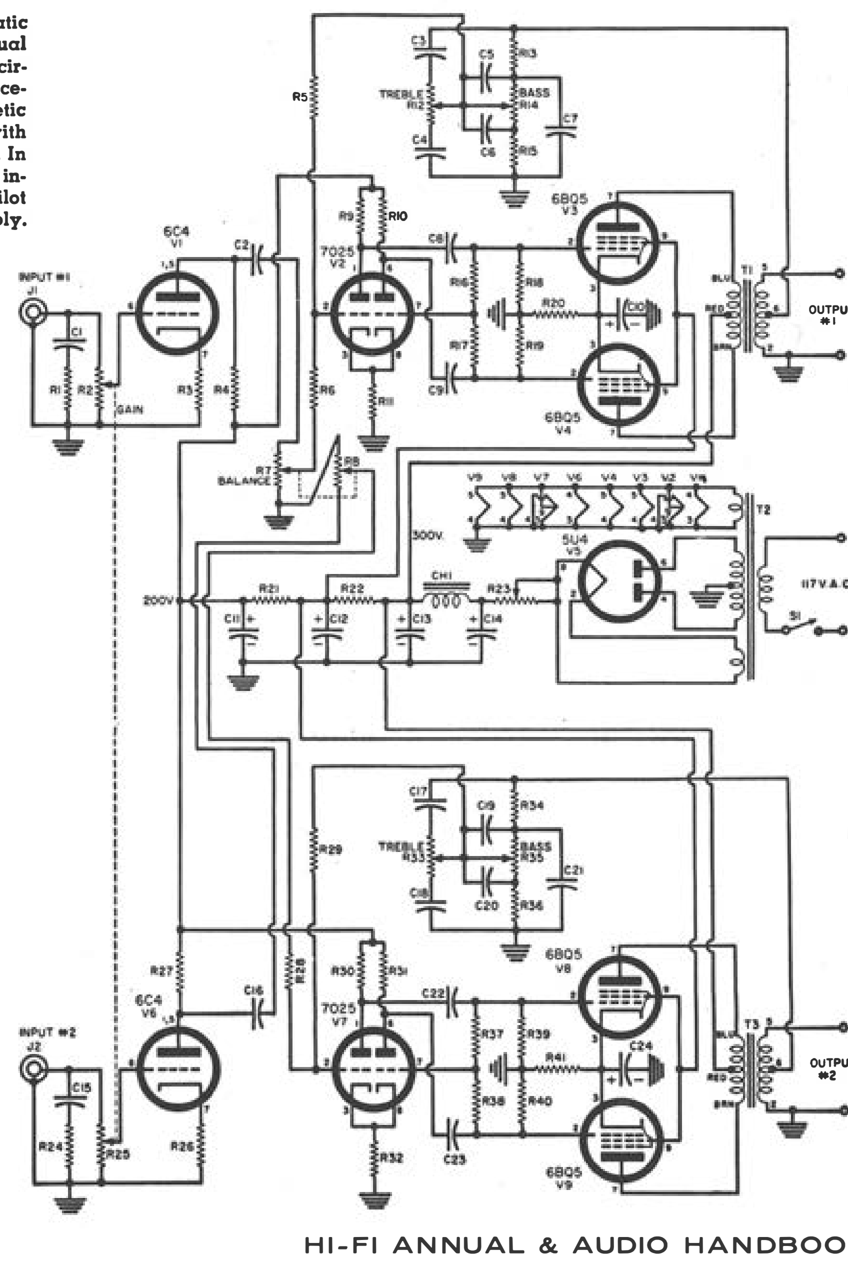


Fig. 3. Here is the complete schematic diagram and parts listing for the dual 10-watt stereo power amplifier. The circuit is designed to accommodate a ceramic stereo cartridge. If a magnetic cartridge is to be used, a preamp with proper equalization would be needed. In this case the RC networks across the input jacks must be removed. A 6-volt pilot lamp may be wired across heater supply.

R_1, R_{24}—180,000 ohm, ½ w. res.
R_2, R_{25}—1 megohm, ½ w. dual linear-taper pot
R_3, R_{26}—1500 ohm, ½ w. res.
R_4, R_{27}—150,000 ohm, ½ w. res.
R_5, R_{29}—1 megohm, ½ w. res.
R_6, R_{28}—470,000 ohm, 1 w. res.
R_7, R_8—500,000 ohm, ½ w. dual linear-taper pot
R_9, R_{10}, R_{30}, R_{31}—270,000 ohm, ½ w. res.
R_{11}, R_{32}—1000 ohm, 1 w. res.
R_{12}, R_{14}, R_{33}, R_{35}—50,000 ohm, ½ w. linear-taper pot
R_{13}, R_{34}—10,000 ohm, ½ w. res.
R_{15}, R_{36}—680 ohm, ½ w. res.
R_{16}, R_{17}, R_{18}, R_{19}, R_{37}, R_{38}, R_{39}, R_{40}—1 megohm, ½ w. res.
R_{20}, R_{41}—150 ohm, 2 w. res.
R_{21}—1000 ohm, 2 w. res.
R_{22}—4700 ohm, 2 w. res.
R_{23}—200 ohm, 20 w. adj. res.
C_1, C_{15}—.002 µfd., 600 v. disc ceramic capacitor
C_2, C_3, C_{16}, C_{17}—.01 µfd., 600 v. disc ceramic capacitor
C_4, C_{18}—.05 µfd., 600 v. disc ceramic capacitor
C_5, C_8, C_9, C_{19}, C_{22}, C_{23}—.02 µfd., 600 v. disc ceramic capacitor
C_6, C_{20}—.2 µfd., 200 v. capacitor
C_7, C_{21}—.0003 µfd., 600 v. disc ceramic capacitor
C_{10}, C_{24}—20 µfd., 50 v. elec. capacitor
C_{11}-C_{12}-C_{13}-C_{14}—20/20/10/10 µfd., 450 v. elec. capacitor
S_1—S.p.s.t. switch
J_1, J_2—RCA-type phono jack
CH_1—2 hy., 200 ma. filter choke (Author used old TV choke. Merit C2974 or equiv.)
T_1, T_3—Universal output trans. 4000/7000/8000/10,000/14,000 ohms c.t. to .17 to 32 sec. (Merit A-2904 or equiv.)
T_2—Power trans. 350-0-350 v. @ 200 ma.; 5.0 v. @ 3 amps; 6.3 v. @ 4 amps (Triad R-20-B or old TV transformer can be used)
V_1, V_6—6C4 tube
V_2, V_7—7025 tube
V_3, V_4, V_8, V_9—6BQ5 tube
V_5—5U4GB tube
Spkrs—6" x 9" ovals (Author used Lafayette SK75)

in Fig. 3. Balance between channels is achieved by the control at the extreme right, R_7, R_8 in the front view. Separate bass and treble controls are used for each channel, at the author's preference, in order to retain flexibility and experiment with intentional tone unbalance of the two channels. However, the individual may prefer single bass and treble controls. If so, he may replace R_{12}, R_{33}, R_{35}, and R_{14} with dual 50,000-ohm pots. To properly adjust the amplifier, set all controls except "gain" at mid-range and set "gain" at a very low level. Using a tone test record (stereo if available, but monaural will do) plug in one input and adjust bass and treble for flat output (or accent highs or lows if preferred). Then plug in the other input and remove the first, adjusting the other bass and treble. Now check the volume from each channel to see if they are equal. If possible do this with a tone record input and an a.c. meter across the voice coil. If this is not feasible, judgment by listening will suffice temporarily. Adjust the balance control until the two outputs are equal. Now over-all gain of the system can be adjusted with the master gain control. There is a possibility that the two sections of the pot used for this control may not have equal resistance throughout their entire ranges, resulting in system unbalance at certain gain settings. About the only solution here is to try another pot, or be content to rebalance the amplifiers at these points. Theoretically, if bass or treble is readjusted, both channels should be changed by the same amount. It has been interesting, however, to experimentally unbalance the tone controls and observe results on various records.

Note that the balance control provides full range from zero to full output for each channel. This, of course, results in a loss of over-all available gain. The author prefers this system since there is a great surplus of gain and full range is desired to experiment with effects and to demonstrate with one channel cut off. If less flexibility and more gain is desired, simply change R_7, R_8, to 100,000 ohms and add 470,000-ohm resistors in series with the low side of R_7, R_8 to ground. This will allow variation in gain of each channel of about ± 20% and will nearly double the preamplifier gain. Such gain is unnecessary and is, in fact, unusable unless a lower output cartridge and more powerful speakers are used, but it is mentioned here to clarify the design.

The Speakers

The speaker system consists of two 6- by 9-inch oval speakers in conventional bass reflex cabinets built into opposite ends of the wall, as shown in Fig. 4. Anyone using an automobile rear-seat speaker will verify that the oval speakers sound pretty good and they proved to be satisfactory in this application. The cost of sound-absorbing insulation in the speaker cabinets was saved by stapling egg cartons of the soft paper variety to the walls.

If you choose small low-cost speakers, remember the amplifiers deliver 10 watts output at full volume. Keep your volume control down to a reasonable level to avoid ruining speakers which may be rated at only 5 watts.

Speaker Placement

Proper phasing of the speakers is obtained by listening for maximum sound reinforcement midway between the two speakers and reversing the leads to one of them, if necessary, to obtain this reinforcement. Improper phasing will leave a "hole" in the music at this central point. A monaural record is helpful in checking for proper speaker phasing.

Referring again to the room diagram of Fig. 4, it is now believed by some that the speakers should be aimed straight out from the wall—not at 45° angles as was once thought. The room is a 12- by 14-foot family room, panelled in knotty pine—a good reflector of the highs. An excellent stereo effect is achieved in most of the room as indicated in the diagram. A "listening test" of the system was made by several friends. Besides being highly pleased with the stereophonic sound, they commented that the panoramic effect when playing monaural records make this unit sound better than most single-channel high-fidelity systems they had previously heard.

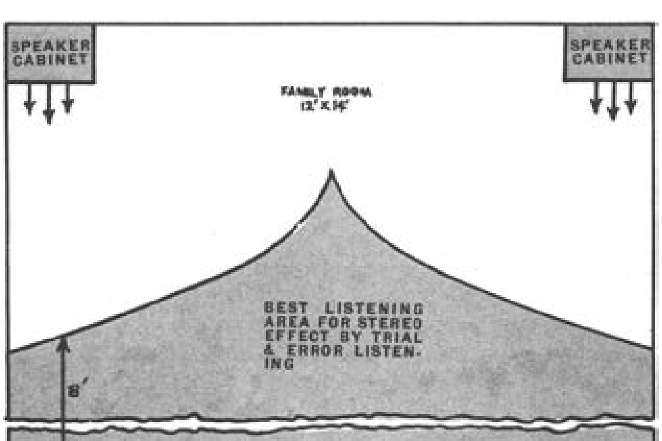
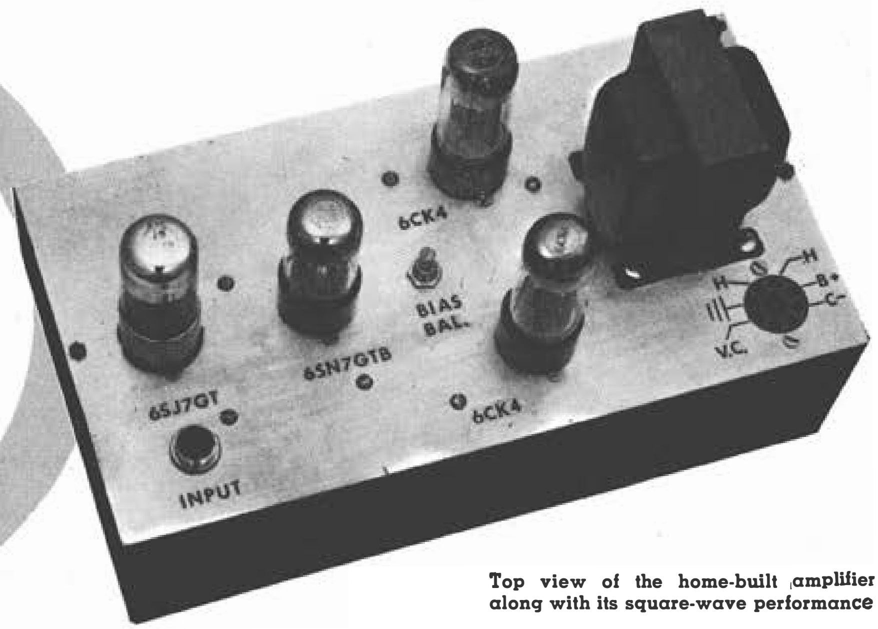
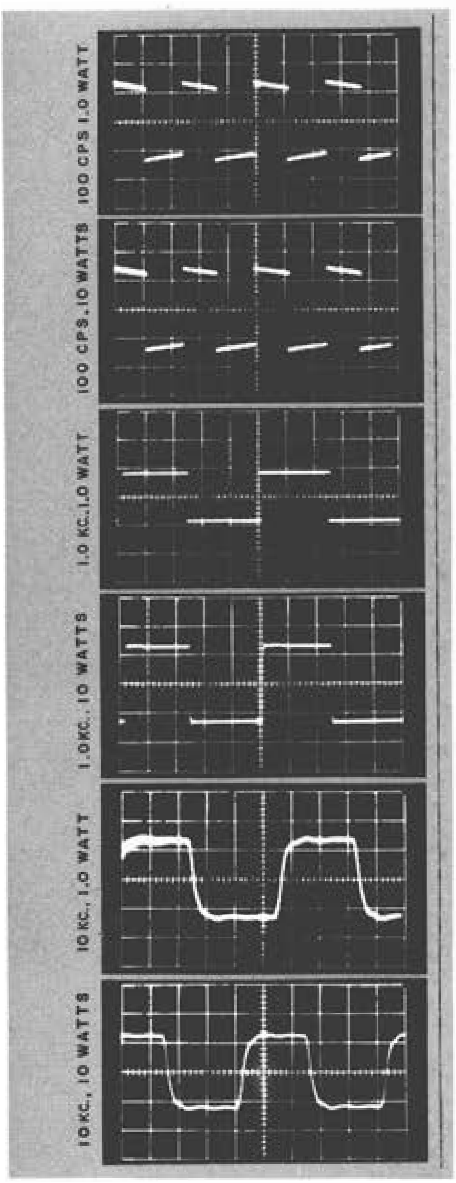


Fig. 4. The speakers are placed along the 12-foot wall of the 12x14 foot room. The area for the best stereo listening is shown shaded in the illustration here.



Top view of the home-built amplifier along with its square-wave performance



Hi-Fi with Triodes

By J. N. STILL

Construction of a 15-watt amplifier with new 6CK4's, which offer the advantage of triode operation plus good efficiency and improved driving requirements.

FROM the dearth of commercial hi-fi amplifiers employing triodes in the output stage, it is obvious that the triode's popularity has been waning in this application. Although fidelity can be achieved with simple circuitry and inexpensive components, the high driving requirements of previously available power-output types imposed additional design problems that could not be overcome economically.

This is not the case with the recently introduced 6CK4. This new cathode-type tube offers all the advantages of triode operation plus good efficiency and improved driving requirements. With the 6CK4 at his disposal, the home builder can now construct a high-performance amplifier at reasonable cost. While the primary purpose of this article is to present and discuss a new amplifier design, let's review briefly the advantages of triode operation.

Advantages of Triodes

Among the "plus" features triodes have to offer the amplifier builder are:

More Uniform Response: An output transformer presents a varying load to the output tubes throughout the audio-frequency range, thus affecting the frequency response of this stage. The low-frequency response may be degraded by the low inductive reactance of the transformer primary winding while the high-frequency response is influenced by the distributed capacity of the transformer. The low effective plate resistance of power-output triodes minimizes these undesired effects, thereby extending and improving uniformity of frequency response.

Damping Characteristics: The load impedance of a loudspeaker varies considerably over the audio-frequency range, with the greatest variations occurring at or near mechanical resonant points. The wide excursions in load impedance due to mechanical resonances are especially objectionable since they cause what is generally known as transient or hangover distortion. This speaker characteristic is sometimes evidenced by excessively high output at particular low frequencies. In order to reproduce transients faithfully and minimize transient or hangover distortion, additional damping must be provided. This is obtained, to some degree, by the internal damping offered by the speaker and the plate resistance of the output tubes. Obviously, only a limited amount of damping can be obtained with carefully controlled loudspeaker design. Since additional damping is required, it becomes evident that one logical way to obtain it is to provide a low plate-resistance-to-load-resistance ratio through the use of triodes.

Distortion and Inverse Feedback: The transfer curve of an output-type triode is normally fairly linear, thereby minimizing amplitude and harmonic distortion. It can also be seen that phase distortion is greatly reduced since the attributes of triode output tubes contribute to wide, uniform frequency response.

Inverse feedback lessens many of the problems that plague the designer of audio amplifier systems. However, applying large amounts of feedback to insure good performance can give rise to economic and design liabilities. Depending on the amount of feedback used, a more expensive output transformer may be required to maintain the necessary degree of stability. The low effective plate resistance of triode power amplifiers and the many benefits derived therefrom, permit the use of less inverse feedback and provides a cost reduction in favor of the designer, while still maintaining top-notch performance.

CLASS AB_1—PUSH PULL	I *	II	III	
Plate Voltage	340	400	400	Volts
Grid Voltage	−43.5	−55	−55	Volts
Grid Voltage r.m.s. Signal	30.8	38.2	38.2	Volts
Zero Signal Plate Current	76	60	60	Ma.
Max. Signal Plate Current	124	106	112	Ma.
Load Resistance	5000	7000	6500	Ohms
Power Output **	15	18.2	18.6	Watts
Total Harmonic Distortion	7.8	8.7	8.85	Per-cent
Plate Dissipation (no signal)	25.8	24	24	Watts
Plate Dissipation (with signal)	27.2	24.2	26	Watts
Efficiency	36.8	43	41.5	Per-cent

* Operation not recommended under these conditions since plate dissipation rating is exceeded.

** Measured directly at tubes. Does not reflect output transformer losses.

Table 1. Typical operation data for a pair of 6CK4's in class AB_1 push-pull.

Triode-Output Amplifier

The circuit diagram of a complete triode-output amplifier is shown in Figs. 1 and 2. The design features push-pull 6CK4's, preceded by the popular duo-triode long-tailed phase inverter and a pentode first stage. This combination is capable of furnishing a healthy 15 watts output to the load with only .7 volt input.

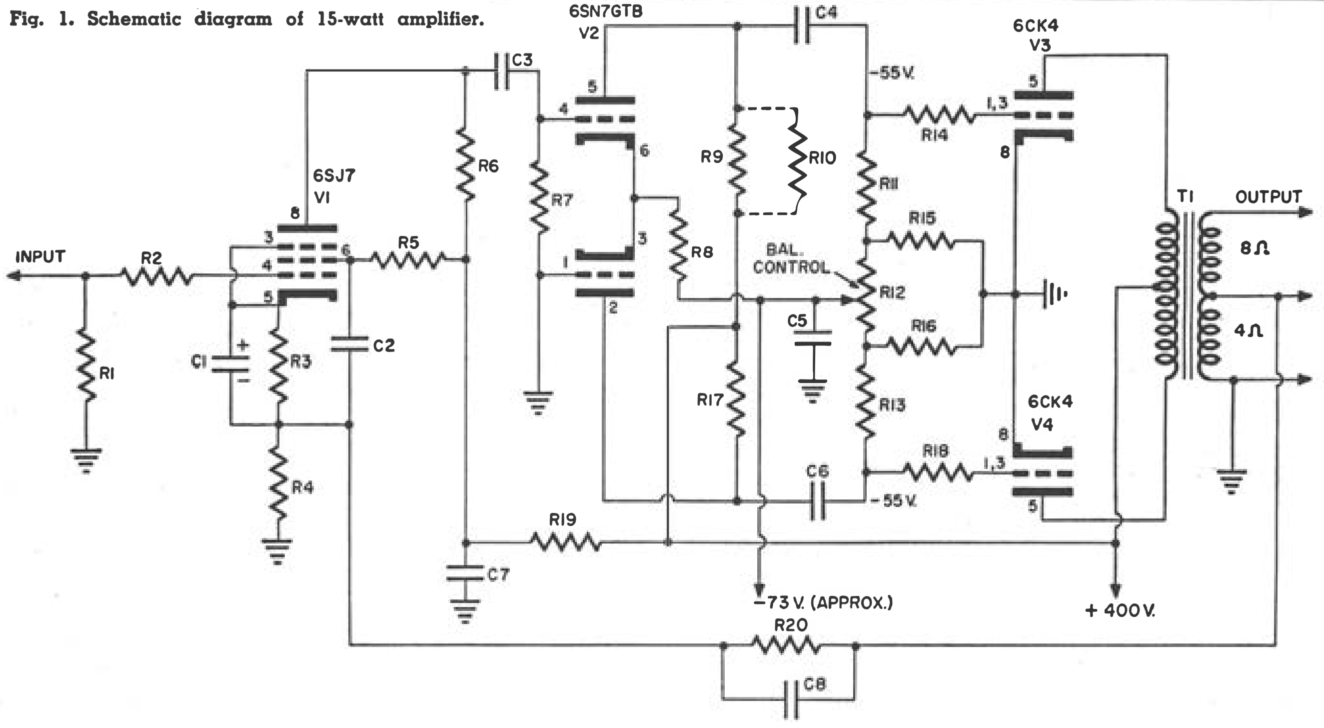
While miniature tubes will perform as well, the Type 6SJ7GT is used for the first stage of the amplifier. The circuit constants of this stage were carefully chosen to provide minimum distortion and maximum frequency response at the input-signal level required to drive the amplifier to full rated output.

The long-tailed phase inverter was selected because it offered reduced phase shift and made possible an extremely stable feedback loop by eliminating a stage of amplification. The tail, or cathode circuit of the inverter, is connected to the negative side of the bias supply to obtain the greatest possible output voltage from this stage. Among other advantages, this arrangement provides a means of bleeding the bias supply while avoiding the direct-coupled circuit configuration usually employed with the long-tailed inverter. (Direct coupling is a convenience used when all factors involved are relatively constant, such as in commercially manufactured equipment. However, in an amplifier that is to be constructed by the home builder, where parts selection is not controllable, particularly the output transformer, direct coupling may introduce problems.)

The possibility of instability due to low- or high-frequency phase shift is minimized by returning the grid of the cathode-driven section of the inverter tube direct to ground. Only the slightest degree of phase imbalance is evident between the two outputs of the inverter. Low-impedance, direct-cathode coupling also contributes greatly to this characteristic. Frequency response of the inverter, without feedback, is essentially flat through 60,000 cps.

If minimum distortion is to be

Fig. 1. Schematic diagram of 15-watt amplifier.



R_1, R_5, R_7—470,000 ohm, 1 w. res.
R_2, R_3, R_{14}, R_{18}—3300 ohm, 1 w. res.
R_4—100 ohm, 1 w. res.
R_6, R_{11}, R_{13}, R_{15}, R_{16}—82,000 ohm, 1 w. res.
R_8—7500 ohm, 1 w. res.
R_9, R_{17}—22,000 ohm, 2 w. res.
R_{10}—150,000 ohm, 1 w. res. (see text)
R_{12}—50,000 ohm pot ("Balance Control")
R_{19}—150,000 ohm, 1 w. res.
R_{20}—1200 ohm, 1 w. res.
C_1—50 μfd., 25 v. elec. capacitor
C_2, C_5—.25 μfd., 400 v. capacitor
C_3—.05 μfd., 400 v. capacitor (see text)
C_4, C_6—.5 μfd., 400 v. capacitor
C_7—16 μfd., 450 v. elec. capacitor
C_8—370 μμfd., 400 v. capacitor
T_1—Output trans. 6500 ohm pri. to 4, 8, 16 ohm sec. (Thordarson T-22S68 or equiv.)
V_1—6SJ7GT tube
V_2—6SN7GTB tube
V_3, V_4—6CK4 tube

RATINGS (Design-Maximum Values)	
D.C. Plate Voltage	550 volts
Plate Dissipation	12.0 watts
Average Cathode Current	100 ma.
Peak Cathode Current	350 ma.
Grid Circuit Resistance Self Bias	2.2 megohms
AVERAGE CHARACTERISTICS	
Plate Voltage	250 volts
Grid No. 1 Voltage	−26 volts
Plate Current	55 ma.
Transconductance	6500 μmhos
Amplification Factor	6.7
Plate Resistance (approx.)	1000 ohms
Grid Voltage for I_b = .5 ma.	−50 volts
Plate Current at E_c = −38 volts d.c.	10 ma.
Zero-Bias Plate Current E_b = 100; E_c = 0 (Instantaneous Values)	125 ma.

Table 2. Here are the maximum ratings and characteristics of one of the new 6CK4's.

FREQUENCY (cps)	TOTAL HARMONIC DISTORTION (%) 1 watt	10 watts	15 watts
40	.65	.7	1.0
50	.6	.8	.9
100	.6	1.0	.8
1000	.5	.5	.6
2000	.45	.7	.8
5000	.8	.8	.8
7500	.8	.85	.9
10,000	1.0	.95	2.0
15,000	1.0	1.6	4.0

Table 3. Total harmonic distortion for the amplifier described in the text.

achieved with limited feedback, care must be taken to insure that each stage of the amplifier exhibits a flat frequency response without inverse feedback. This condition is absolutely essential if feedback is to be used to reduce distortion rather than as a means of extending frequency response. This principle of relatively flat frequency response with little feedback was one of the important considerations in developing the design of the first stage and the phase inverter.

The only drawback to the long-tailed phase inverter is the possibility of unequal output voltages due to the slightly higher gain of the grid-driven section. This condition can be corrected by reducing the value of load resistance used in the plate circuit of the grid-driven half of the inverter until the outputs are of equal amplitude. As shown in Fig. 1, balance is obtained by shunting the specified load resistance with different value resistors (R_{10}) starting with 150,000 ohms.

Output Stage

The output tubes operate class AB_1, under conditions shown in the third column of Table 1. The two additional columns of data are presented for purposes of comparison. Table 2 summarizes the published rating and characteristics of the Type 6CK4. Fixed bias is employed rather than self-bias in the output stage mainly to obtain those few extra watts of power that would otherwise be lost. A 5000-ohm potentiometer, incorporated in the bias supply, provides a means of adjusting the voltage appearing at the grids of the 6CK4's to the proper operating level of −55 volts, Fig. 2. Static plate-current balance is obtained by adjusting the 50,000-ohm "Balance Control" connected in the grid circuits of the output tubes, Fig. 1. Balance is obtained when a minimum voltage differential exists between the plates of the output tubes. As an alternate system, a 5-ohm sampling resistor could be built into the cathode circuit of each 6CK4.

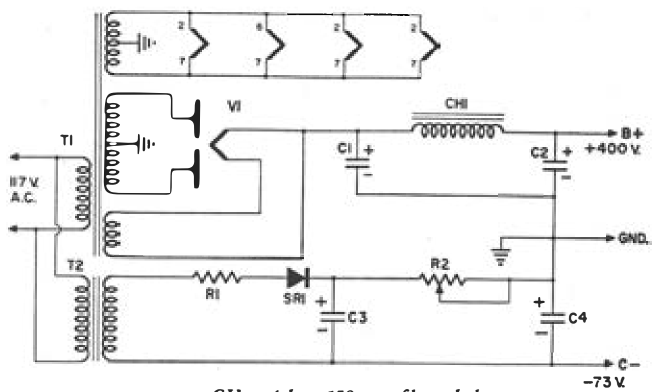
Earlier in the article it was pointed out how a triode output stage eliminates many of the problems that lead to more complex circuitry and the incorporation of a relatively expensive output transformer. These savings are, of course, reflected in the amplifier described. The transformer used to couple the 6CK4 output stage to the speaker system is a *Thordarson* Model T-22S68, currently selling for about $6.50

What remains to be said about the output stage can best be covered by describing the over-all amplifier performance.

Performance

When considering an amplifier for home use, it is important to keep in mind the fact that a high average power rating is sought only to insure that the amplifier can provide the high peak power required for faithful reproduction. Actual average power outputs in excess of a few watts are rarely needed in the home and an average output of around 10 watts will more than satisfy all but extreme conditions. The amplifier described is capable of furnishing 35 watts peak power, at the load, thus comparing favorably with the peak power capacity of the best 25-watt pentode amplifiers. This is quite an impressive feat when the fact that the amplifier has an average power-output rating of 15 watts is considered.

Fig. 2. Complete schematic diagram and the parts values for the associated power supply to be used with the 15-watt amplifier.



R_1—22 ohm, 1 w. res.
R_2—5000 ohm, 4 w. pot
C_1—16 μfd., 500 v. elec. capacitor
C_2—16 μfd., 450 v. elec. capacitor
C_3, C_4—20 μfd., 150 v. elec. capacitor

CH_1—4 hy., 150 ma. filter choke
SR_1—25 ma. selenium rectifier
T_1—Power trans. 400-0-400 v. @ 150 ma.; 6.3 v. @ 3.5 amps; 5 v. @ 3 amps
T_2—Bias trans., 117 v. @ 25 ma.
V_1—5U4GB tube

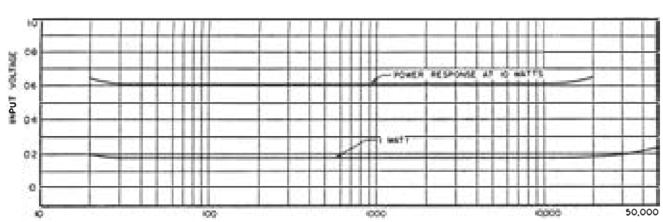


Fig. 3. Power response curves of the amplifier taken at 1 and 10 watts output.

Power-response curves at 1 and 10 watts are shown in Fig. 3. The amplifier is virtually flat within the range of 30 to 16,000 cps at 1 watt output. At 10 watts output the amplifier is essentially flat from 40 to 14,000 cps. The power sensitivity of the amplifier is also apparent from the power response curves, *i.e.,* approximately .6 volt drive will produce 10 watts output. The amplifier's maximum average power output of 15 watts is obtained with only .7 volt input. The excellent power sensitivity is achieved by employing only 12 db of inverse feedback, which is a direct reflection of the combined benefits of the triode output stage.

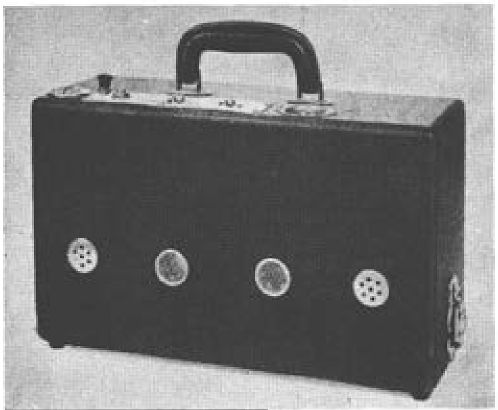
Total harmonic distortion at 1-, 10-, and 15-watt levels is shown in Table 3 for the range of 40 through 15,000 cps. At 1-watt output the distortion is 1 per-cent or less through 15,000 cps. The particular output transformer employed causes the distortion to rise above 1 per-cent at 15,000 cps at the 10- and 15-watt levels. Although of little consequence, since the power level of program material seldom exceeds a few watts above 12,000 cps, the latter condition can be corrected by selecting a different transformer.

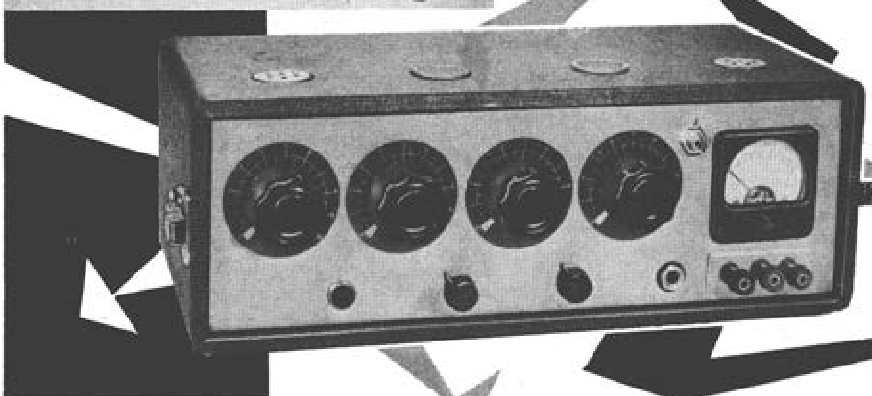
The excellent linearity of the amplifier is also shown by the low percentage of intermodulation distortion present. With frequencies of 400 and 1000 cps, in a 4:1 ratio, the intermodulation distortion is only 1 per-cent at 1.5 watts and .8 per-cent at 15 watts.

The unretouched square-wave oscillograms shown further attest to the capabilities of the amplifier. The 100- and 10,000-cps square waves exhibit slight low- and high-frequency roll-off, while the 1000-cps square wave is reproduced almost perfectly. Although a somewhat more square 10,000-cps wave was obtained with a smaller value capacitor in the feedback loop, the larger value used in the final design assures ring-free performance at the higher power levels. No high-frequency ringing was in evidence with the amplifier reproducing the 10,000-cps square wave even when a .1-μfd. capacitor was shunted across the output transformer secondary—indicating good high-frequency stability.

The amplifier is also extremely stable at low frequencies despite the use of somewhat oversized coupling capacitors. Should low-frequency instability be encountered, due possibly to the use of an output transformer other than the specified model or a power supply without adequate filtering, the value of the coupling capacitor between the plate of the 6SJ7 and the phase inverter, C_3 should be reduced slightly. The exceptional stability of the amplifier is further indicated by the absence of oscillation with the output transformer plate-leads incorrectly phased to provide positive instead of negative feedback. As a result of this characteristic, the builder must determine the output transformer primary connection that provides negative feedback. Of the two connections possible, the correct connection will be characterized by lower amplifier sensitivity.

In conclusion, the performance specs speak for themselves. Here is a high-quality, medium-power amplifier that can be built at low cost and with remarkably few construction problems.





Audio "Mix-It" Box

By **LEON A. WORTMAN**

Operating panel of the "Mix-It" box is shown here. With cover in place, unit is portable.

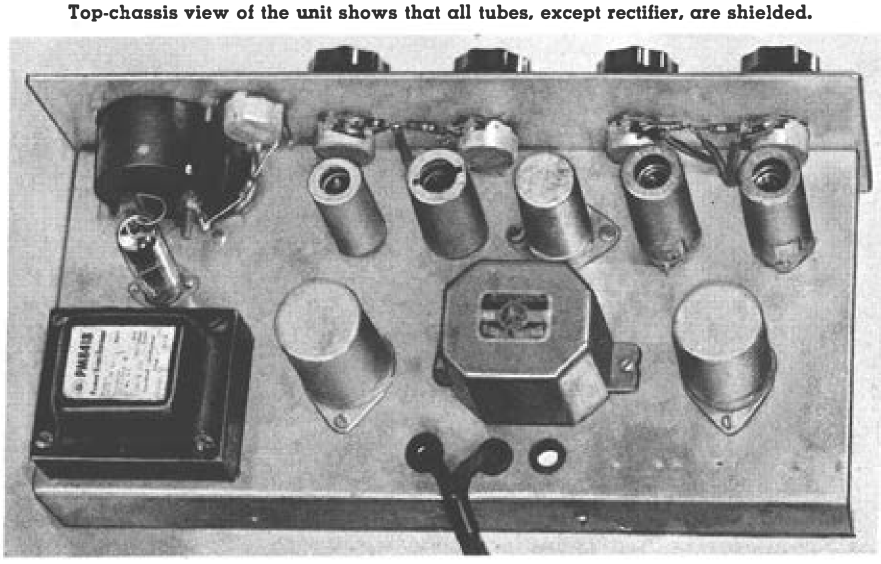
This versatile mixer-preamplifier will handle four separate audio signal sources at one time.

IF THE signal is at an audio frequency, this versatile mixer-preamplifier can handle it very nicely. In fact, it will "handle" four separate audio signal sources at one time. With it, the operator can mix or blend or balance such diverse signals as those from these sources: low-impedance microphone (50 to 250 ohms); high-impedance microphone (above 500 ohms); reluctance-type phono cartridge; and utility (bridging) from any impedance source. Yes, microphones, tuners, tape players, phonographs, oscillators, or radio lines—high or low gain, high or low impedance—whatever the source, the "Mix-It" provides an excellent means of controlling signal level with both visual and aural monitoring.

Included in the circuit are: a db meter and meter attenuator, a phone jack, output terminals, plus all manual controls mounted on the front panel for easy handling and viewing. The output terminals are three thumbscrew binding posts. One is common and isolated from the chassis, the other two are for connection to an 8-ohm voice coil and a 500-ohm line. All input connectors are at rear of cabinet.

Power is provided by a built-in transformer-type a.c. supply. The entire "Mix-It" box measures only 14" wide, 8" deep, and 5" high when closed in its carrying case. You see, it is portable too! Total weight is a mere 7 pounds.

Top-chassis view of the unit shows that all tubes, except rectifier, are shielded.



The chassis is a standard 13" x 7" x 1½" open-end aluminum unit. The cabinet is a fabric-covered wood box obtained from a local photo-supply store at very low cost. The box was originally designed as a carrying case for a 35 mm slide projector. The inside dimensions proved ideal for the standard chassis being used. The panel was cut from aluminum stock to fit the front opening of the carrying case. Other cases can be used, of course, in conjunction with standard-size aluminum or steel chassis. The panel is secured to the chassis by the hex nuts of the "on-off" switch, pilot light, master gain control, and the hardware of the output terminal strip. For an attractive and distinctive appearance, after the panel had been drilled and cut, the panel was sprayed with gold lacquer. One of the handy spray cans, available at radio parts houses and many hardware stores, was used for the purpose. Other colors, such as the familiar grays and blacks, and hammertone sprays are also available. Their application is simple and certainly enhance the appearance of the equipment.

The tube line-up and functions are as follows, referred to the schematic diagram of Fig. 1: V_{1A} preamplifies the low-impedance microphone-level signals from input transformer T_1; V_{1B} preamplifies the signals of the high-impedance microphone or crystal-type

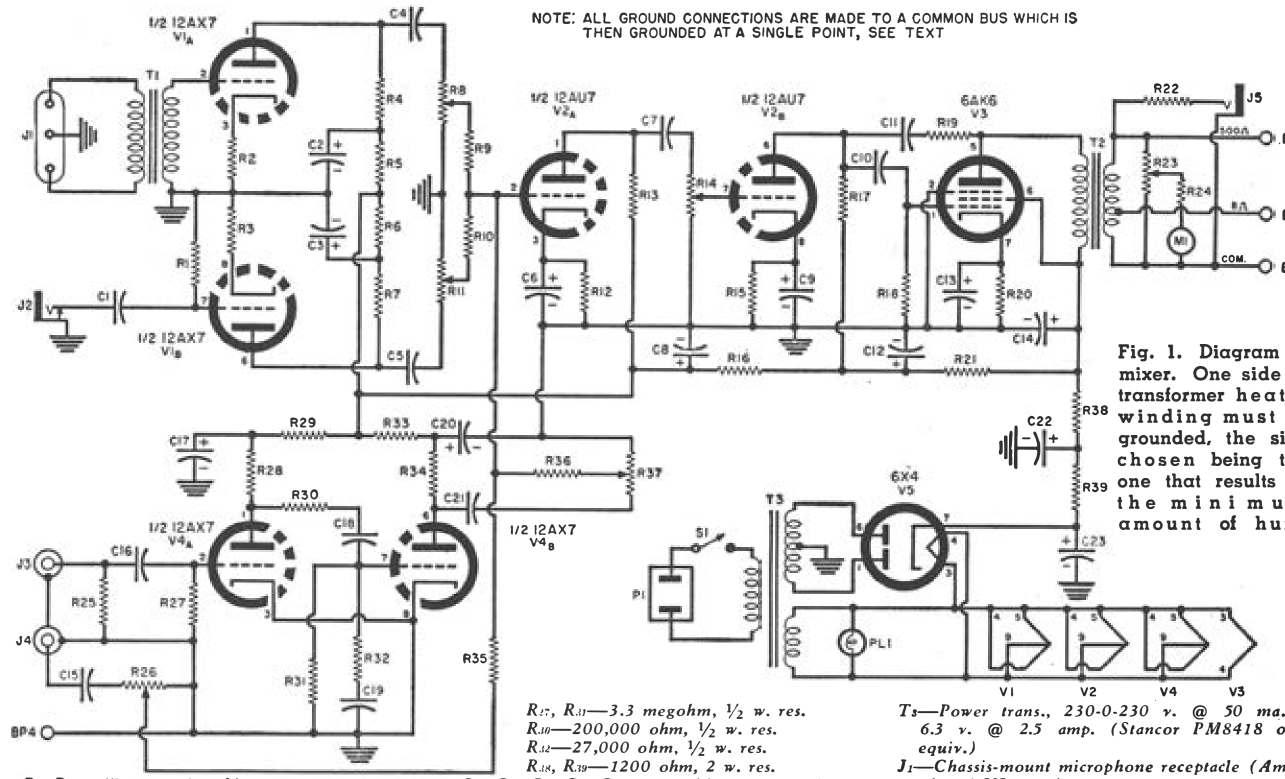


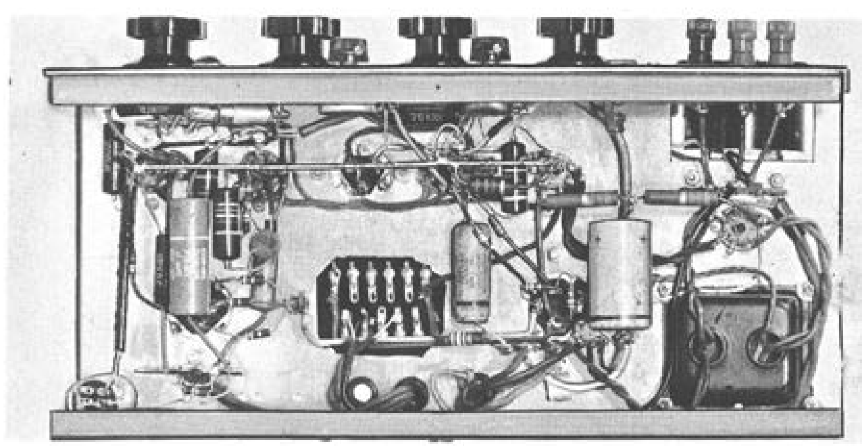
Fig. 1. Diagram of mixer. One side of transformer heater winding must be grounded, the side chosen being the one that results in the minimum amount of hum.

R_1, R_{18}—470.000 ohm, ½ w. res.
R_2, R_3, R_{15}, R_{22}—1500 ohm, ½ w. res.
R_4, R_7, R_9, R_{10}, R_{35}, R_{36}—100,000 ohm, ½ w. res.
R_5, R_6, R_{33}, R_{34}—33,000 ohm, ½ w. res.
R_8, R_{11}, R_{14}, R_{26}, R_{37}—500,000 ohm pot
R_{12}—2200 ohm, ½ w. res.
R_{13}, R_{17}, R_{28}, R_{29}—68.000 ohm, ½ w. res.
R_{16}, R_{21}, R_{24}—3900 ohm, 1 w. res.
R_{19}—39,000 ohm, 1 w. res.
R_{20}—560 ohm, ½ w. res.
R_{23}—10,000 ohm pot
R_{25}—15,000 ohm, ½ w. res.
R_{27}, R_{31}—3.3 megohm, ½ w. res.
R_{30}—200,000 ohm, ½ w. res.
R_{32}—27,000 ohm, ½ w. res.
R_{38}, R_{39}—1200 ohm, 2 w. res.
C_1, C_4, C_5, C_7, C_{21}—.02 µfd., 400 v. tubular capacitor
C_2-C_3-C_{17}-C_{20}, C_8-C_{12}-C_{14}-C_{22}—10/10/10/10 µfd., 450 v. elec. capacitor
C_6-C_9-C_{13}—20/20/20 µfd., 25 v. elec. capacitor
C_{10}, C_{16}, C_{18}—.05 µfd., 400 v. tubular capacitor
C_{11}, C_{15}—.1 µfd., 600 v. tubular capacitor
C_{19}—.01 µfd., 400 v. tubular capacitor
C_{23}—16 µfd., 450 v. elec. capacitor
T_1—Microphone trans., low impedance to grid (UTC O-1 or equiv.)
T_2—Universal output to line trans., 10,000 ohms to 500 ohms (UTC S-15 or equiv.)
T_3—Power trans., 230-0-230 v. @ 50 ma.; 6.3 v. @ 2.5 amp. (Stancor PM8418 or equiv.)
J_1—Chassis-mount microphone receptacle (Amphenol XL-3-13)
J_2—Closed-circuit phone jack
J_3, J_4—Phono pin jack
J_5—Open-circuit phone jack
BP_1, BP_2, BP_3, BP_4—Binding post
P_1—A.c. receptacle (Amphenol 61M10)
S_1—S.p.s.t. switch (author used rotary unit)
PL_1—6.3 v. pilot light
M_1—Decibel meter
V_1, V_4—12AX7 tube
V_2—12AU7 tube
V_3—6AK6 tube
V_5—6X4 tube

phono cartridge; V_{4A} and V_{4B} preamplify and bass-compensate the output of a reluctance-type phono cartridge; V_{2A} and V_{2B} are in series-connected circuits as voltage amplifiers driving V_3, a 6AK6 power pentode. Maximum audio power output available is approximately .8 watt into a 500-ohm load; V_5 is a 6X4 full-wave rectifier tube for the power supply.

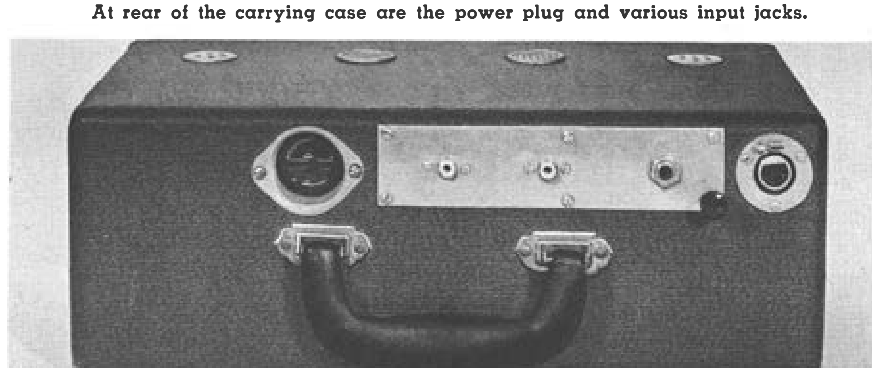
The "Mix-It" box was originally constructed for use with a remotely located tape recorder of professional quality, therefore the db meter, meter attenuator, and multiple outputs were required. These features are not often found in equipment other than that designed for commercial recording or broadcast work. The db meter and meter attenuator enable the "setting of levels" when the operator of the "Mix-It" and the recorder are physically separated. This permits the adjustment of volume controls during a "feed" or a "pickup" for optimum performance with respect to distortion (by controlling possibilities of overloading) and background noise (by maintaining a relatively high minimum level of the audio signal consistent with good dynamic range). Frequency response of the "Mix-It" is shown in Fig. 2.

The output level of the "Mix-It" is



Bottom-chassis view of the audio "Mix-It" unit. Note use of a ground bus.

At rear of the carrying case are the power plug and various input jacks.



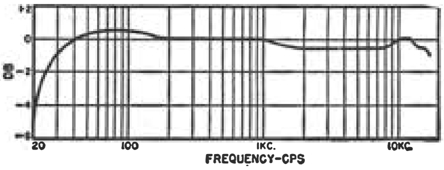


Fig. 2. Over-all response curve of mixer.

sufficient to drive a small monitor loudspeaker connected to the 8-ohm output terminals. This permits the equipment to be used for a number of additional purposes. It can be used as a playback amplifier for tapes or discs, as an entertainment amplifier for listening to a radio tuner or record player, or to monitor the signal during an "off the air" recording, or to monitor tape or disc dubbing sessions.

The input connections are mounted on a small strip of metal at the back of the box. The low-impedance microphone receptacle is an *Amphenol* type frequently used with such equipment. The high-impedance microphone plugs into a standard normally closed, single-circuit phone jack. The inputs for the reluctance cartridge and the bridging connector are pin-type phono jacks. A thumb-screw binding post provides a convenient means of connection to the common ground bus, sometimes found necessary for minimizing "ground loops." Note that the ground is *not* carried through the output terminals. All cables from the input terminal strip to the chassis are flexible and shielded for long life and minimum hum pickup from this source. The a.c. line cord plugs into the back of the cabinet. When the "Mix-It" is not in use or is being transported, the power cord can be folded up and stored inside the cover of the cabinet.

With care in wiring, hum and oscillation should not present a problem. Filament wiring, as is good practice, should be done first and should "hug" the chassis. A common ground bus, No. 12 tinned wire, is run about ½" in space above the pins of the audio tubes, from V_1 to V_4. All ground connection for these stages should be made to the ground bus, with the cathode and grid resistors terminated at one point for each stage. The ground bus is connected to the chassis at one point only. Experiment will quickly locate that point which gives best results, that is, minimum hum. A small screwdriver, placed against the ground bus and the chassis, is moved along until the optimum grounding point is indicated. In this particular construction that point was determined to be at the approximate center of the ground bus, where the three-section electrolytic, C_8-C_9-C_{13}, is mounted. Vent plugs are mounted at the top of the cabinet directly above the tubes. In addition to providing air flow, they enable rapid visual inspection of the tubes with respect to filament glow, just in case some trouble should develop.

Storing YOUR Small Components

By FORREST H. FRANTZ, SR.

Keep your small items in corrugated "parts cards."

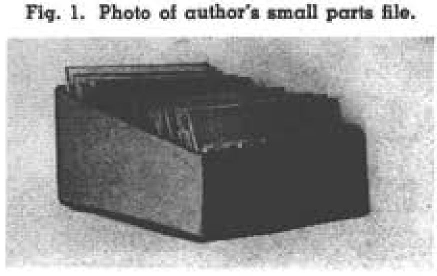
THE serious transistor experimenter has a problem. His problem is storing transistors and miniature parts. Other problems associated with this one are: (1) what kind and how many of each part does he have on hand and (2) what are the basic characteristics of transistors of a certain type?

The problem is not peculiar to the experimenter. The matter of control is even more important to radio and TV technicians, laboratory stock room personnel, and distributors. A system of control which hinges on physically seeing the parts in stock is ideal—if the parts are assembled into a physically small space and counting can be done rapidly. A very good answer to the problem occurred to the author, triggered by *Sylvania* which shipped transistors stuck into corrugated paper sheets such as those often used between layers of candy and cookies in commercial packages. The author tried to obtain some of this corrugated paper locally but couldn't find any in a hurry.

A decision was then made to improvise "parts cards" out of corrugated cardboard cut from shipping boxes and pasted on pieces of stiff cardboard which served as backing and provided writing space for component identification and characteristic information.

These "parts cards", as employed by the author in this case, vary in size. This was done to facilitate location. A rubber band around the card helps to keep resistors and other "two ended" parts such as capacitors in place. With small components stored on cards like this, they can be filed in a very small space, all conveniently separated and tagged. The author's small parts file shown in Fig. 1, can hold over 100 transistors, about 400 resistors, about 100 small ceramic capacitors, and approximately 100 small electrolytic capacitors for transistor work without cramping. The box is only about 6 inches wide by 9 inches long!

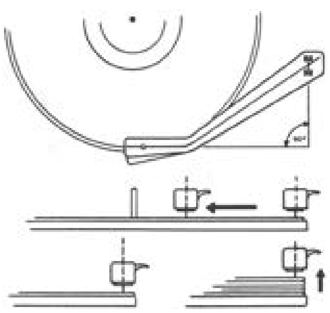
Fig. 1. Photo of author's small parts file.



To know the difference . . . is to select the DUAL-1006 combination stereo turntable/changer

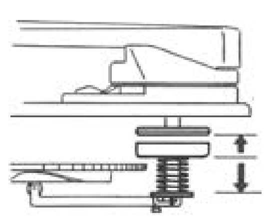
united audio
PRODUCTS OF DISTINCTION
202-4 East 19th St., N.Y. 3, N.Y.

PRECISION MOTOR — Exceptionally powerful motor ensures *constant* RPM; enables turntable to reach full speed in less than ¼ turn from dead start! New, exclusive *rigid-equipoise* motor suspension, eliminates rumble at its source; retains original dynamic balance, prevents bearing wear. *Total* shielding and location of motor in relation to tonearm make hum pickup impossible.



Always perfect vertical and lateral stereo tracking because arm pivot axis remains 90° to cartridge axis.

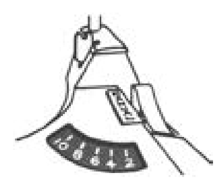
PROFESSIONAL TONEARM — *One-piece* construction avoids multiple arm resonance or distortion of cartridge output. Snap-in 4-terminal cartridge holder inserts and locks easily, quickly into tonearm head, takes all standard size stereo and mono cartridges — even turnover and turnaround types. Double set of ballbearings (in both axes) provide *complete* freedom of lateral and vertical motion.



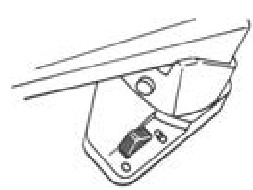
TRACKS AS LOW AS 1½ GRAMS — Jam-proof *clutch* completely frees tonearm from the cycling cam during play; engages the arm only during cycling action. This eliminates drag and prevents tracking force variations, preserves original cartridge compliance specifications; even permits *automatic* and *changer operation* at this low pressure!

BIG TURNTABLE — New principle of laminated and concentrically-girded construction defies warping and eccentricity. Especially deep axle shaft prevents wobble, and is imbedded in heavy duty ball bearing assembly. Though of heavy armor-gauge, slim streamlined appearance is achieved by recessing hi-torque inner drive rim.

DRIVE GEARS DISENGAGE AUTOMATICALLY — Self-locking and trouble-free, a multiple transmission is used, with a set of *individual* gears for each of the four speeds (16⅔, 33⅓, 45, 78 rpm). All drive gears and idler *totally* disengage when motor is off; no need to adjust speed control, no "neutral" position to remember, never any "flat spot thump."



BUILT-IN PRESSURE GAUGE — A vital necessity for the preservation of valuable records and for optimum cartridge operation. *Direct reading* and conveniently located for instant check of stylus tracking force. *Wide range* tonearm weight adjustment faces forward for fingertip accessibility.

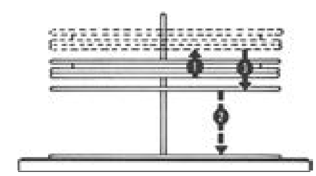


MULTI-ACTION STEREO/MONO SWITCH — Not only instantaneously adapts cartridge for stereo or mono output, but also *removes* random noises produced by mono records when used with stereo cartridges. *No* switching at amplifier necessary for mono or stereo reproduction.

TRUE "TURNTABLE" OPERATION MANUAL OR AUTOMATIC SINGLE PLAY
Three choices with the 1006! Start manually, with either a rotating or *motionless* turntable, setting the tonearm anywhere on the record to play all or just a desired portion. Or push a button and the 1006 starts, and finds the record lead-in groove. In all cases, the tonearm returns to rest after play, motor shuts off, and drive disengages . . . all *automatically!* Stop, repeat, or reject (manually or automatically) any time you please. You can even *interrupt* play, and start again where you left off! Short, single-play spindle provided.

AUTOMATIC CHANGER — Whether for stereo or mono reproduction, the outstanding qualities of the 1006 as a professional turntable are remarkably preserved as a record changer too! Look . . .

SIMPLE CONTROLS — Simple to operate in spite of its many special features — only three buttons start, stop, reject, or repeat action. Repeat button is self-cancelling after replay, or can repeat same record any number of times without disturbing the stack. Spindle need not be removed from turntable to remove records or rearrange record sequence . . . even while record is playing.



1. *Record stack separates from bottom record.* 2. *Bottom record descends.* 3. *Stack gently lowers for next play.*

ELEVATOR ACTION SPINDLE — To prevent wear to record center holes, spindle *lifts* entire stack except bottom-most record, allowing it to descend to turntable freely and effortlessly. Then stack is gently lowered — not dropped — into position for next change. Customary stabilizer guides and pusher arms thus eliminated; no offsets in spindle to chip records.

OBSOLESCENCE-PROOF INTERMIX — Patented roller-feeler guide in tonearm head permits stacking up to ten records in *any* sequence and of any diameter from 5" to 12"! No worry about future record sizes; solves problems of variations in "standard" records.

CONSTANT CHANGE-CYCLE SPEED — Uses independent auxiliary transmission for the automatic cycling action, avoiding needless strain and wear on the playback gears, and provides rapid change *regardless* of the record speed setting or turntable load.

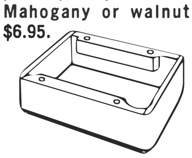
QUIETING CIRCUITS — Self-muting and squelch filter circuits keep the electrical operation of the 1006 as wonderfully quiet as the mechanical action of the skillfully created assembly. No "pops" or buzzes.

Model 1006 — Supplied with standard 3½ lb. turntable, single play and changer spindles, 2 cartridge holders, adapter disc for 45's, instructions, template. Chassis: 10¾" x 13" — clearances: 3" below, 6" above. 110/220v., 60 cy. A.C. (50 cy. available). Price: $69.95 net.

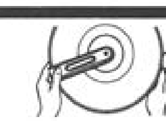
Model 1006X — same as 1006, but with 5½ lb. turntable. $74.50 net.

Pickering 371 — Mark II Stereo Cartridge with **special** high compliance diamond stylus, mounted and wired to cartridge holder. Order separately or with either model Dual. $26.40 net.

Model WB — Wood Base pre-cut, ready to install. Mahogany or walnut. $6.95.



Model AS-6 45 RPM Automatic Spindle, for playing stack of ten 7" records without center hole inserts. $4.80.



Model RC-1 Record Cleaner — Cleans both sides of record at once. Specially impregnated to lubricate and preserve grooves. Operates **in** direction of grooves, never across, preventing damage. Removes static charges and annoying noises. $2.95.

Automatic Motor Control For Hi-Fi Turntables

By **GEORGE F. ANDREWS**

Sound-operated relay, with adjustable delay, turns off phono motor if audio is missing for more than 15 seconds.

MOST owners of manual hi-fi turntables, while well satisfied with the wowless, rumble-free operation obtained, have probably encountered the same problem as the author, *i.e.,* after placing a 12-inch LP album on and settling back to enjoy the pleasant sounds, a sudden distraction such as the telephone, children's squabble, wife, etc., calls you away.

Thinking you will surely return before the album is finished, you depart. Forty-five minutes later, you suddenly remember that the machine is still running and you hurry back to find the needle grinding its life away on the last groove. Having repeated this scene numerous times, the device to be described evolved. Before the final design was adopted, the possibility of using a photocell circuit to shut off the turntable was investigated; but, due to the difference in location of the final groove on different sizes and brands of recordings, it was ruled out, and the following system was devised.

In effect, it is a sound-operated relay, with an adjustable delay time, which is set to turn off the turntable motor if the audio signal is missing for more than fifteen seconds. This allows the device to distinguish the pauses between selections on an album from the actual end of the recording. One toggle switch is the only operational control and its function is to select either the *automatic* feature or normal *manual* operation of the turntable motor.

One of the main requirements was that the control should involve no direct connections to the amplifier, since a true hi-fi addict will tolerate nothing which might load or alter the response of his equipment. Thus, it was decided to use "air coupling," the necessary signal being picked up by a four-inch PM speaker placed inside the existing enclosure, at the same time insuring that extraneous room sounds do not prevent the relay from operating.

A transistor preamp was desirable, but the use of batteries in any form was not, since the unit should be able to be installed and then require no further attention. Therefore, the transistor stage is operated from a simple internal power supply. The switching is done by a 2D21 miniature thyratron with d.c. on its plate so that, once fired, it will remain conducting until the operator restarts the motor.

Use is made of standard, inexpensive parts throughout and after three months of daily use, operation is still positive and without fault. Referring to the diagram, circuit operation is as follows: When the hi-fi system is playing, a small signal is picked up by the PM speaker "mike" and applied through T_1 (an ordinary a.c.-d.c. radio audio output transformer, 2500 ohms-to-voice coil, connected in reverse) to the base of the transistor. This stage is used in the grounded-emitter configuration which results in a low input impedance for a correct match to T_1. Circuit constants are chosen so that limiting occurs in this stage in order to present a constant level output to the following stage, thus insuring that the delay time remains the same for different listening levels. The transistor output is coupled through T_2 (a standard 1:3 midget interstage audio transformer) to the diode and thence to the *RC* circuit in the grid of the thyratron.

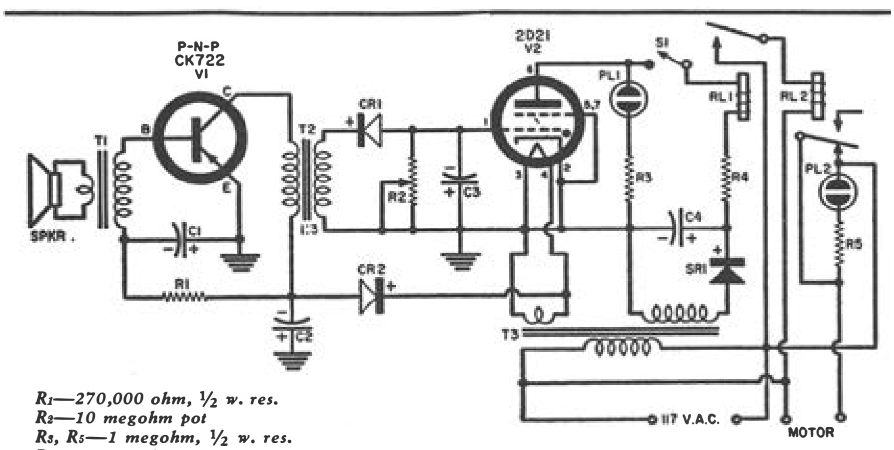
Here the amplified, limited signal is rectified by CR_1 and negatively charges C_3 to hold V_2 cut off. Should the incoming signal cease, C_3 starts to discharge through R_2 (which is adjustable to cover a range of zero to approximately 35 seconds delay) and when the grid potential decays to -2 volts, the thyratron conducts and closes RL_1, in turn operating RL_2, which stops the turntable motor. V_1 remains conducting until S_1 is opened, thus restoring the circuit to its original condition and restarting the motor.

The power supply is a conventional half-wave selenium type, and the transistor "B−" (*p-n-p* transistor) is obtained by rectifying the 6.3-volt thyratron filament supply with a 1N34 diode and its associated filter capacitor. Almost any available plate relay can be used for RL_1, since the limiting factor would be the current capability of the power supply used. With the 125-volt supply shown, the 15,000-ohm plate load resistor limits the current to approximately 15 ma., which is the rating of the midget power transformer used. The two NE-2 lamps are for pilot lamps only and are not necessary for proper circuit operation. Wiring is not critical and the unit may be built with ease on a 2" x 4" x 6" chassis.

After construction the first adjustment is the proper placing of the pickup speaker in the enclosure to insure adequate signal at normal listening level to limit in the transistor stage. This can be observed with an oscilloscope at the collector of the CK722, or with a v.t.v.m. on the grid of V_2. If the latter method is used, the pickup speaker should be placed close enough to the hi-fi speaker so that increasing the volume no longer increases the bias. The final adjustment is with R_2 to set the delay time. Fifteen seconds has been found best to cover all types of musical selections without premature operation of the relay.

The switching circuit of this device could easily be altered to turn off the whole system or perhaps control some other useful function. Its adaptability to other equipment will be left to the ingenuity of the reader.

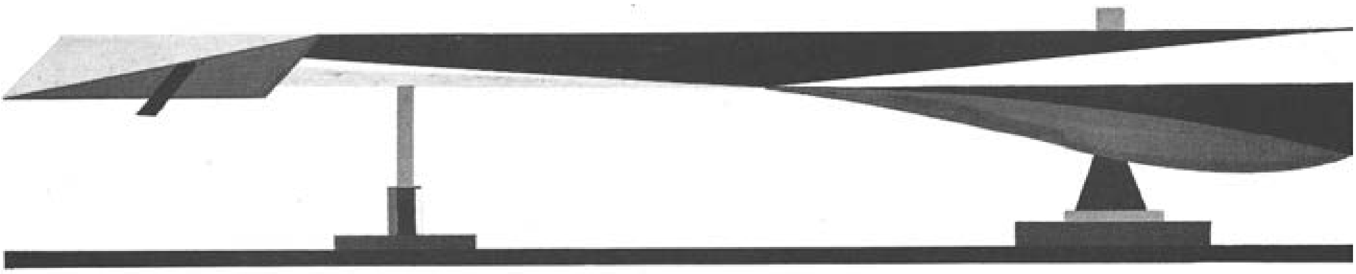
Complete schematic diagram and parts listing for the automatic motor control unit.



R_1—270,000 ohm, 1/2 w. res.
R_2—10 megohm pot
R_3, R_5—1 megohm, 1/2 w. res.
R_4—15,000 ohm, 1 w. res.
C_1—10 µfd., 25 v. elec. capacitor
C_2—100 µfd., 25 v. elec. capacitor
C_3—1000 µfd., 15 v. elec. capacitor
C_4—40 µfd., 150 v. elec. capacitor
CR_1, CR_2—1N34 germanium diode
SR_1—50 ma. selenium rectifier
PL_1, PL_2—NE-2 neon bulb
RL_1—S.p.s.t., normally open, plate relay (see text)
RL_2—S.p.d.t. relay, 110 volt a.c. coil (Leach 1118)
S_1—S.p.s.t. toggle switch
Spkr.—4" PM speaker
T_1—Audio output trans., midget 2500 ohm-to-v.c. type
T_2—Audio interstage trans., 1:3 ratio, midget type (Merit A-53-C)
T_3—Power trans. 125 v. @ 15 ma.; 6.3 v. @ .6 amp., midget type (Stancor PS-8415)
V_1—"p-n-p" junction transistor (CK722)
V_2—2D21 tube

Transistorized Phono Preamp for Stereo

By FRANCIS A. GICCA/Raytheon Co.



Construction of high performance, low-cost preamplifier for magnetic stereo cartridges that affords the maximum response. Design features minimum shunting capacitance.

THE most critical components in a stereophonic disc reproducing system are the stereo cartridge and its associated preamplifier. Unless these two components deliver maximum performance, the finest amplifiers and speakers cannot hope to produce true stereophonic fidelity. If a stereo cartridge can inherently reproduce the entire audio spectrum with high fidelity, then it is obvious that its preamplifier must not destroy this fidelity or add hum, noise, or distortion of its own. This article will cover the development of a preamplifier designed to achieve optimum performance with magnetic stereo cartridges.

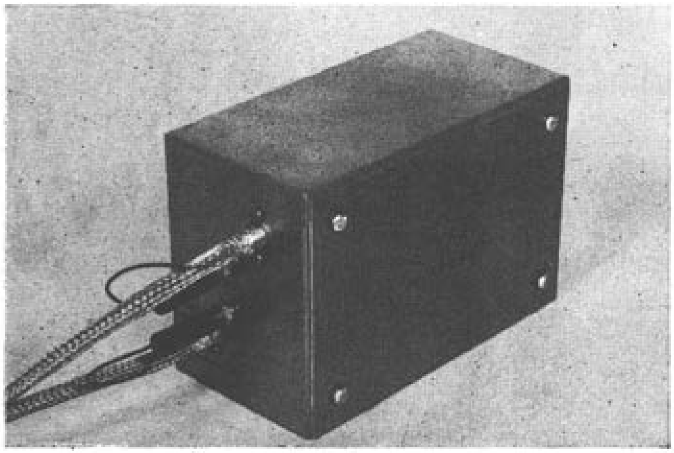
By their very nature, magnetic stereo cartridges have low outputs in the 1 to 10 millivolt range. At 60 cps, RIAA equalization requires a bass boost of 16 db. This bass boost, coupled with the stereo cartridge's low output, can easily lead to serious hum in the preamplifier unless the initial design overcomes this inherent problem. The obvious solution is to completely isolate the preamplifier from a.c. fields by either using a highly filtered d.c. heater supply in a vacuum-tube preamp or by eliminating tubes and using transistors. The latter approach is preferable since transistors are inherently hum-free and the expense of a low-ripple d.c. heater supply is avoided. Therefore, it was decided that the best low-hum preamplifier could be built using transistor circuits.

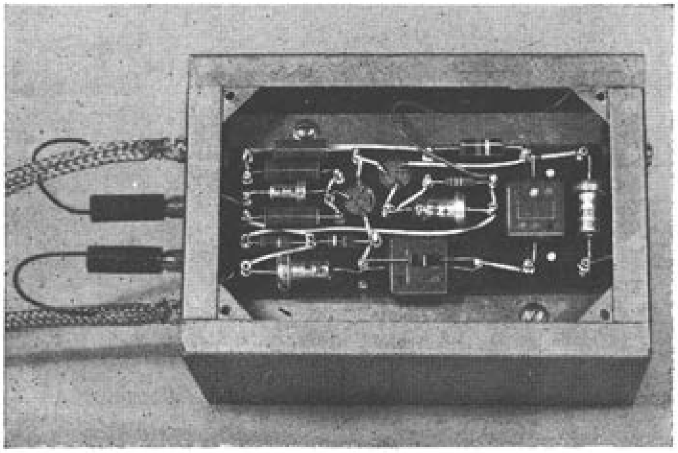
In order to extract the smoothest and widest frequency response from a magnetic cartridge, it is vital that the preamplifier offer optimum loading for the cartridge. Unfortunately, a basic conflict exists in trying to load a magnetic cartridge properly by resistive means alone.

Fig. 1A shows the basic equivalent circuit for a magnetic cartridge, its cabling to the preamplifier, and the preamplifier loading. At frequencies below cartridge resonance, shunt capacities may be neglected, yielding the equivalent circuit of Fig. 1B. This circuit is essentially a low-pass RL filter whose response is down 3 db at $f = (R_c + R_L) / 2\pi L_c$. In order to prevent a droop in response in the audio band, this corner frequency must be above the highest frequency it is desired to reproduce. This, in turn, requires that $R_c + R_L$ be greater than $2\pi L_c$ at this frequency, which requires a large value of R_L since R_c is fixed by the cartridge chosen and is generally a low value. Unfortunately, a high value R_L can cause excessive thermal noise in the preamplifier, generated by thermal agitation in this large resistor. This leads to the first conflict; a large value R_L for minimum low-pass filter effects, but a small value of R_L for minimum thermal noise generation.

Consider the high-frequency resonant equivalent of Fig. 1C. Here the cartridge appears like a resonant RLC circuit which exhibits the resonant response shown. If R_L is large, then the resonant peak will be correspondingly large. If the resonant peak occurs in the audio band, due to high cable and preamplifier capacitance, the resonant peak should be made as small as possible through the use of a low value R_L in order to keep the response smooth. If at all possible, the resonant peak should be kept out of the audio band since cartridge output drops off rapidly above resonance. These facts suggest that the value of R_L should be kept low, as should cable and preamplifier capacitance. This, again, is in conflict with the requirement of a large value of R_L for minimum low-pass

Over-all and internal views of the preamplifier. The interior view, taken with one of the chassis side panels removed, shows the underside of one of the two wiring boards used in the construction.





filter effects. For an excellent discussion of cartridge loading see "Loading the Phono Cartridge" by Herman Burstein.

Because of these conflicts in a choice of loading resistance, many cartridge manufacturers recommend a value of loading resistance which compromises low-pass and resonance effects. Fortunately, there is a solution to this dilemma which is not a compromise and affords maximum response from all cartridges.

If cable and preamplifier capacitance can be reduced to only one or two micromicrofarads, then cartridge resonance will occur well above the audio spectrum and R_L can be increased indefinitely since the resonant peak will be inaudible. A large value of R_L then places low-pass filter effects outside the audio band.

However, R_L must be kept physically small in order to minimize thermal noise. This appears to conflict with the foregoing but it need not if a physically small resistor is *electronically* multiplied in value to appear like a large resistor. Fig. 2A shows how this may be accomplished. Transistor V_1 is connected as an emitter-follower, which is roughly equivalent to a vacuum-tube cathode-follower. If the emitter-follower has an a.c. gain of 0.9, then the a.c. output voltage (E_o) across R is 0.9 E_i, the input voltage. An impedance Z is connected between input and output. Summing voltage drops around the input yields: $E_i = IZ + 0.9\ E_i$ or $0.1\ E_i = IZ$. The input impedance seen across terminals 1 and 2 is then $E_i/I = 10Z$. This shows that the effective impedance seen across terminals 1 and 2 is ten times greater than the actual impedance Z. If Z is a resistor, then ten times this resistance is seen across terminals 1 and 2. If Z is a capacitor, then one-tenth its capacitance is seen. If the gain of the emitter-follower is 0.99 instead of 0.9, then 100 times Z is seen at the input.

This is exactly what is desired since effective capacitance is greatly reduced and effective resistance is greatly increased while still maintaining a resistor of small physical size.

Like a cathode-follower, the emitter-follower has a low output impedance. Therefore, terminal 3 of Fig. 2A is close to ground potential. This fact allows the shield of the cartridge cable to be connected to terminal 3 which places the cable capacitance from the input to the output of the emitter-follower, thereby effectively lowering the cable capacity. With an emitter-follower gain of 0.99, a cable capacitance of 100 $\mu\mu$f. is reduced to a negligible value of 1 $\mu\mu$f. Of course, the cartridge output must still be applied between terminals 1 and 2, which requires a third lead to carry the ground. Preferably, this should be a second shield about the cable, but may be a separate unshielded wire, as shown in Fig. 2B.

Fig. 3 is the schematic diagram of the complete preamplifier. In order to prevent duplication of discussion, only the left channel circuit will be covered. What is said will, of course, apply to both channels.

CARTRIDGE	Lc (mhy.)	Cc ($\mu\mu$f.)	Rc (ohms)	3 db Point Low-Pass (kc.)	Resonant Peak* (kc.)	Preamp Output (volts)	Booster Output (volts)
Electro-Sonic C-100	1	Unmeas.	40	9000	2000	.039	.250
Fairchild XP-4	3	Unmeas.	600	3000	1500	.120	.756
Fairchild 232	4	Unmeas.	600	2000	1000	.120	.756
General Electric GC-5	500	50	600	18	30	.240	Note 1
Grado Magnetic	1	Unmeas.	600	9000	2000	.078	.491
Pickering 371		Note 2		Not Calculable		.390	Note 1
Scott 1000		Note 2		Not Calculable		.160	Note 1
Shure M3D	350	30	440	30	45	.200	Note 1
Stereotwin ST-200		Note 2		Not Calculable		.980	Note 1

*Resonant peak calculated includes effect of 500 $\mu\mu$f. cable capacity.
Note 1. Use of booster amplifier with these cartridges not recommended, in order to keep distortion level low.
Note 2. This data not available for these cartridges.

Table 1. Measured characteristics of some typical magnetic stereo cartridges along with their outputs when employing the circuits described in the text.

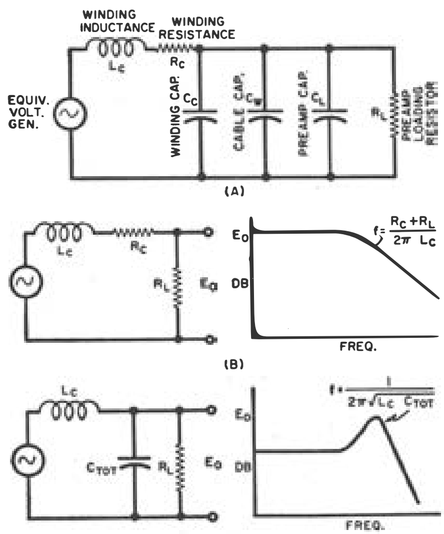
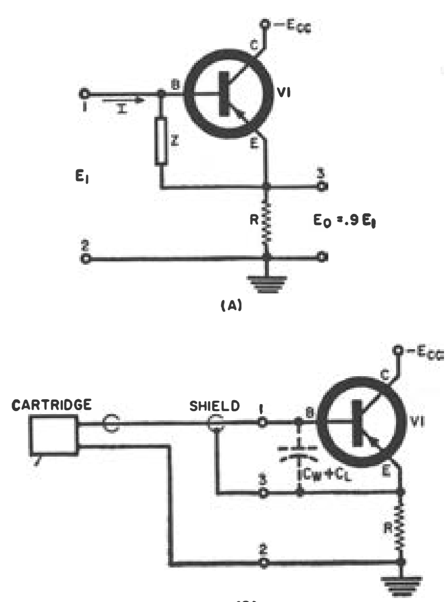


Fig. 1. (A) Cartridge, cable, and preamp equivalent circuit. (B) Equivalent circuit below cartridge resonance. (C) Equivalent circuit at cartridge resonant frequency.

Fig. 2. (A) Emitter-follower used as impedance multiplier. (B) Emitter-follower effectively lowers cable, preamp capacitance.



Note that input transistor V_1 is connected as a conventional common-emitter amplifier and not as an emitter-follower as previously discussed. However, as will be seen, the voltage across unbypassed resistor R_5 is actually a large fraction of the input voltage as previously detailed for an emitter-follower. Transistor V_1 has a common-emitter gain of about 20 db which means that ten times the input voltage appears across the 30,000-ohm collector load resistor, R_3. Since the same a.c. current flows through the 1000-ohm emitter resistor R_5, the voltage across this resistor must to 1/30th the voltage across the collector resistor (1000/30,000) or ⅓rd the input voltage appears across R_5. This leads to an impedance multiplication of 1.5—surely not a spectacular amount.

Feedback accomplishes the remaining impedance multiplication. The feedback network, C_4-R_6-C_8, from the collector of V_2 to the emitter of V_1 serves two purposes. First, it applies RIAA equalization to the preamplifier. Second, it feeds back sufficient output signal to R_5 in the proper phase such that the voltage across R_5 is approximately equal to the input signal. As a result, the impedance multiplication factor is about 80, surely adequate to minimize capacitance and allow the use of a physically small loading resistor for minimum thermal noise. The loading resistance is formed by the parallel combination of R_1 and R_2 (about 18,000 ohms), which establishes bias for V_1.

At low frequencies, the reactance of feedback capacitor C_4 is large. Therefore, at low frequencies negative feedback is small and the impedance multiplication factor is correspondingly small and approaches 1.5. Fortunately, a high value loading resistance is needed only at high frequencies where the low-pass filter effect comes into play. Actually, a lower impedance at lower frequencies is advantageous since the preamplifier offers a lower input impedance to stray hum fields and minimizes their pickup.

Fig. 4 shows the actual measured variation of input impedance with fre-

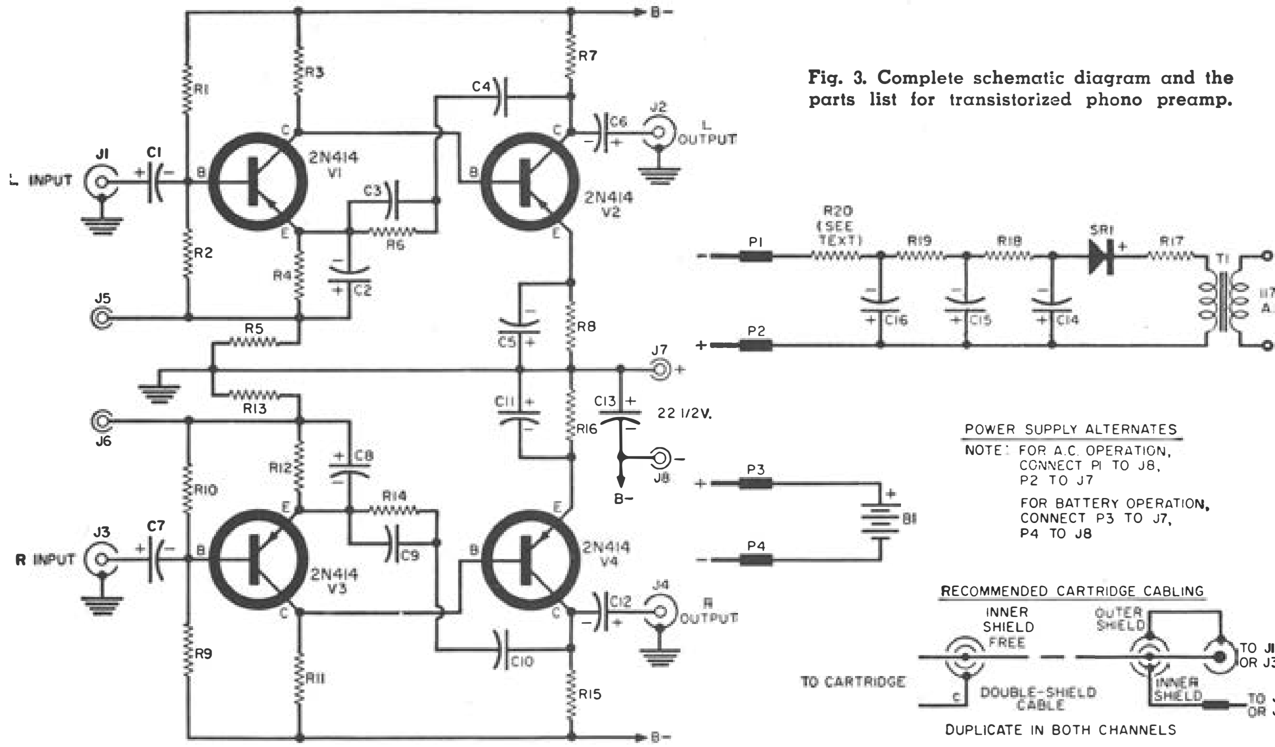


Fig. 3. Complete schematic diagram and the parts list for transistorized phono preamp.

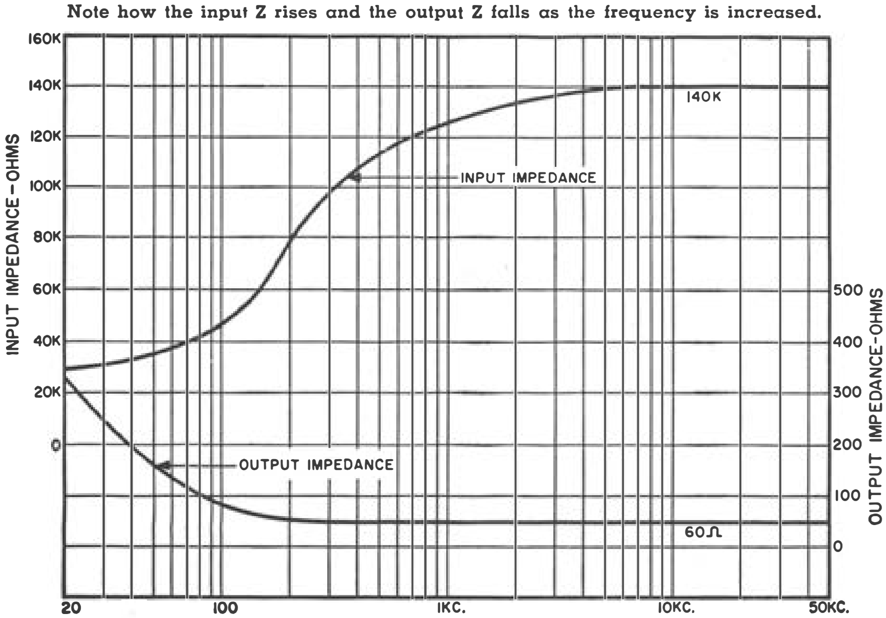
R_1, R_9—240,000 ohm, low-noise deposited carbon res. (see text)
R_2, R_{10}—24,000 ohm, low-noise deposited carbon res. (see text)
R_3, R_{11}—30,000 ohm, low-noise deposited carbon res.
R_4, R_{12}—3900 ohm, 1/2 w. res.
R_5, R_{13}—1000 ohm, 1/2 w. res.
R_6, R_{14}—30,000 ohm, 1/2 w. res.
R_7, R_{15}—15,000 ohm, 1 w. res.
R_8, R_{16}—18,000 ohm, 1/2 w. res.
R_{17}—30 ohm, 1/2 w. res.
R_{18}, R_{19}—27,000 ohm, 1/2 w. res.
R_{20}—See text
C_1, C_7—8 μf., 30 v. elec. capacitor
C_2, C_5, C_6, C_8, C_{11}, C_{12}, C_{13}—40 μf., 30 v. elec.
C_3, C_9—2700 μμf., 500 v. mica capacitor
C_4, C_{10}—6800 μμf., 500 v. mica capacitor
C_{14}—12 μf., 250 v. elec. capacitor
C_{15}, C_{16}—20 μf., 150 v. elec. capacitor
J_1, J_2, J_3, J_4—RCA phono jack
J_5, J_6, J_7, J_8—Pin jack
P_1, P_2, P_3, P_4—Pin plug
SR_1—Selenium rectifier, 20 ma., 150 v. (Federal 1159 or equiv.)
T_1—Isolation trans., 117 v. sec., 35 va. (Triad N51X or equiv.)
B_1—22 1/2-volt battery
V_1, V_2, V_3, V_4—"p-n-p" transistor (Raytheon Type 2N414)

quency. At low frequencies the input impedance is 27,000 ohms which is 1.5 times 18,000 ohms, the parallel combination of R_1 and R_2. At high frequencies the input impedance is 140,000 ohms, which represents a multiplication of 80.

Fig. 4 also illustrates another important virtue of the equalization feedback circuit; that of lowering the output impedance of the preamplifier. At high frequencies the preamplifier's output impedance is a low 60 ohms which allows the use of a longer length of shielded cable to the main power amplifiers without any attenuation of highs.

The preamplifier uses two direct-coupled common-emitter stages because this configuration is least sensitive to bias changes which may be caused by variations in transistor parameters or temperature. The d.c. feedback in the form of bypassed emitter resistors (R_4 and R_8) also stabilizes the preamplifier against transistor changes. The resulting basic preamplifier is very stable and has extremely wide response. As a matter of fact, without feedback equalization, the 1 db-down points of the preamplifier occur at 9 and 340,000 cps. This fact guarantees absolute minimum preamplifier phase shift within the audio band. Unless both preamplifiers have negligible phase shift in the audio band a phase error will exist between channels and cause a severe loss of stereophonic effect. The inherent stability of the preamplifier is indicated by the fact that the gain varies less than one decibel from zero to 85 degrees centigrade.

Fig. 4. Curves showing the variations in the input and output impedances of unit. Note how the input Z rises and the output Z falls as the frequency is increased.



Raytheon Type 2N414 transistors were chosen because of their wide response, low noise, and low cost (only \$2 each). The entire preamplifier can be built for less than \$25 complete. In order to keep the price as low as this without sacrificing performance, the gain of the preamp is average, which means that the gain is adequate for medium- and high-output magnetic cartridges, but insufficient for low-output cartridges. If low-output cartridges are to be used, the gain will have to be increased by the addition of a booster transistor amplifier following the main preamplifier. Fig. 7 is a schematic of an appropriate booster amplifier for low-output cartridges. Clearly, it is an expensive waste to build in gain to handle all cartridges if a medium- or high-output cartridge is to be used.

Table 1 shows what outputs can be

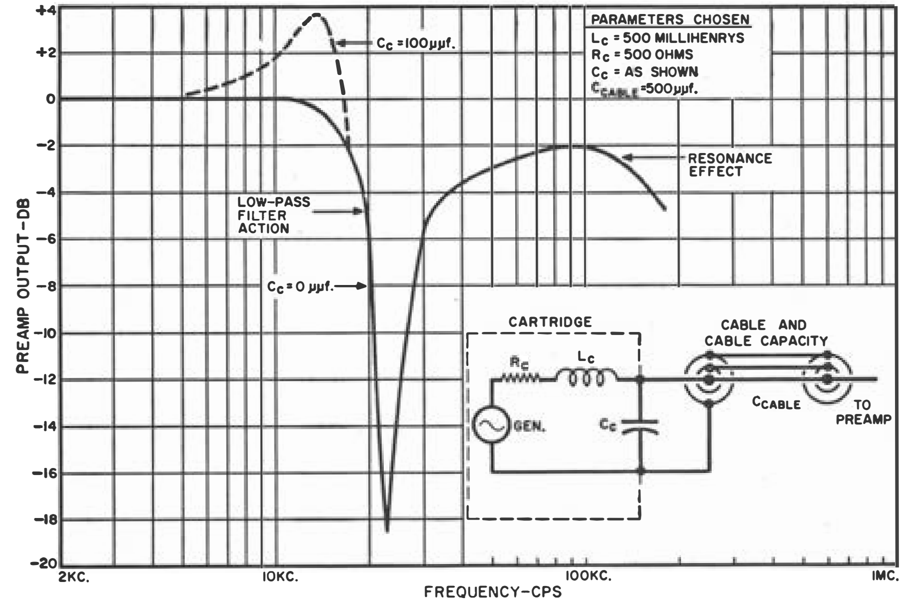


Fig. 5. Curve showing the measured high-frequency effects described in the text.

expected from typical stereo cartridges which were checked with the preamplifier—both with and without the booster. Table 1 also indicates which cartridges should not use the booster amplifier. Medium- and high-output cartridges should not use the booster because the high levels will cause unnecessary distortion in the booster amplifier and the output level from the preamplifier is adequate.

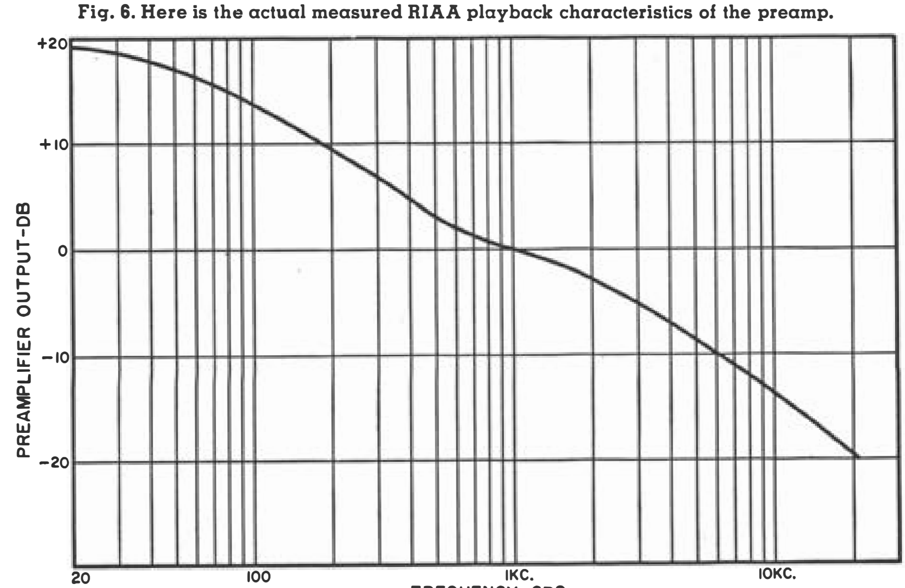
As a further means of reducing cost, tone and volume controls were not incorporated. Actually, such controls do not belong on the preamplifier proper, but rather on the main amplifier or control center. The principal task of the preamplifier is to draw maximum stereo performance from the cartridge and to accurately apply RIAA playback equalization. This the transistor preamp does, and does excellently.

Performance

In order to evaluate the performance of the preamplifier, several tests were run. First, feedback capacitor C_4 was shunted with a large value capacitor (40 µf.) and capacitor C_3 removed. This applies a constant amount of feedback across the audio band and allows analysis on a flat-response basis rather than having to take the RIAA characteristics into account. Such an approach is valid since the RIAA characteristic merely superimposes its own curve on the flat response of the preamplifier. With the preamplifier thus modified, it is flat to 340,000 cps and allows the investigation of high frequency low-pass and resonance cartridge effects.

Fig. 5 shows the test setup used. A cartridge equivalent was constructed as shown and connected to the preamplifier using a 500 µµf. cable—then the preamplifier's response was measured. Low-pass filter and resonance effects will be most severe with large cartridge inductances (L_c). Of all the cartridges checked, the *General Electric* Model GC-5 had the highest inductance —500 mhy —so this value was used.

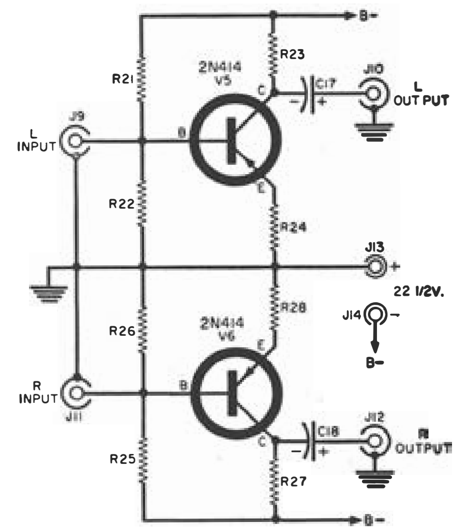
Fig. 6. Here is the actual measured RIAA playback characteristics of the preamp.



The GC-5 has a coil capacitance (C_c) of 50 µµf. which places the cartridge resonance at 30,000 cps. In order to make the problem more severe, 100 µµf. was used which places the resonant peak at 13 500 cps. Note that even with this severe capacitance the response is reasonably flat to 17,000 cps, rising only 3.7 db at resonance. With the actual value of 50 µµf., resonance effects lie completely outside the audio band.

Removing C_c allows evaluation of low-pass filter effects. For the GC-5 inductance of 500 mhy., the preamplifier causes a low-pass 3 db point at 18,000 cps. The resonance shown in Fig. 5 at 95,000 cps is due to cable capacitance reduced in value from 500 µµf. to approximately 5 µµf. by the preamplifier.

The GC-5 represents the most severe



R_{21}, R_{25}—180,000 ohm, ½ w. res.
R_{22}, R_{26}—13,000 ohm, ½ w. res.
R_{23}, R_{27}—15,000 ohm, ½ w. res.
R_{24}, R_{28}—2000 ohm, ½ w. res.
C_{17}, C_{18}—40 µf., 30 v. elec. capacitor
J_9, J_{10}, J_{11}, J_{12}—RCA phono jack
J_{13}, J_{14}—Pin jack
V_5, V_6—"p-n-p" transistor (Raytheon 2N414)

Fig. 7. Schematic of booster amplifier.

case of high-frequency effects due to its high inductance. Table 1 indicates the location of the 3 db low-pass filter point and resonant peak for several other popular cartridges, measured in the same manner as for the GC-5. All cartridges listed are unaffected by the preamplifier and response is limited *only* by the response of the *cartridge* itself. If the GC-5 is to be used, its low-pass 3 db point should be raised to 36,000 cps by increasing R_1 and R_9 to 470,000 ohms and R_2 and R_{10} to 47,000 ohms. This increases the noise level slightly, but this is compensated by the higher output of the GC-5. As a result, the signal-to-noise ratio remains the same as for lower output cartridges.

Feedback equalization was restored and RIAA equalization measured. Fig. 6 shows that the characteristic follows RIAA to better than a decibel. Signal-to-noise ratio with equalization is 70 db and total distortion is less than 0.7% including harmonic and intermodulation terms.

Construction of the preamplifier is straightforward and non-critical. Any type of construction can be used successfully, but it is wise to keep lead lengths short and vital that the preamplifier be totally enclosed in a metal case to shield it from hum fields.

Power for the preamplifier can be obtained either from a battery or a separate negative power supply. A 22½-volt battery is preferable for minimum hum and should last for about six months under continuous use. However, the alternate power supply shown in Fig. 3 can be used as well. This power supply can be built for about $7 and yields a signal-to-hum ratio of over 70 db. The value of power supply resistor R_{20} depends on whether or not a booster is to be used and powered by the power supply. If a booster is *not* used, R_{20} should be a 27,000-ohm, ½-watt resistor. If a booster *is* used, R_{20} should be omitted.

Values of the electrolytics are not critical. If you cannot obtain a listed value, substitute a larger capacitance with the same or higher voltage rating.

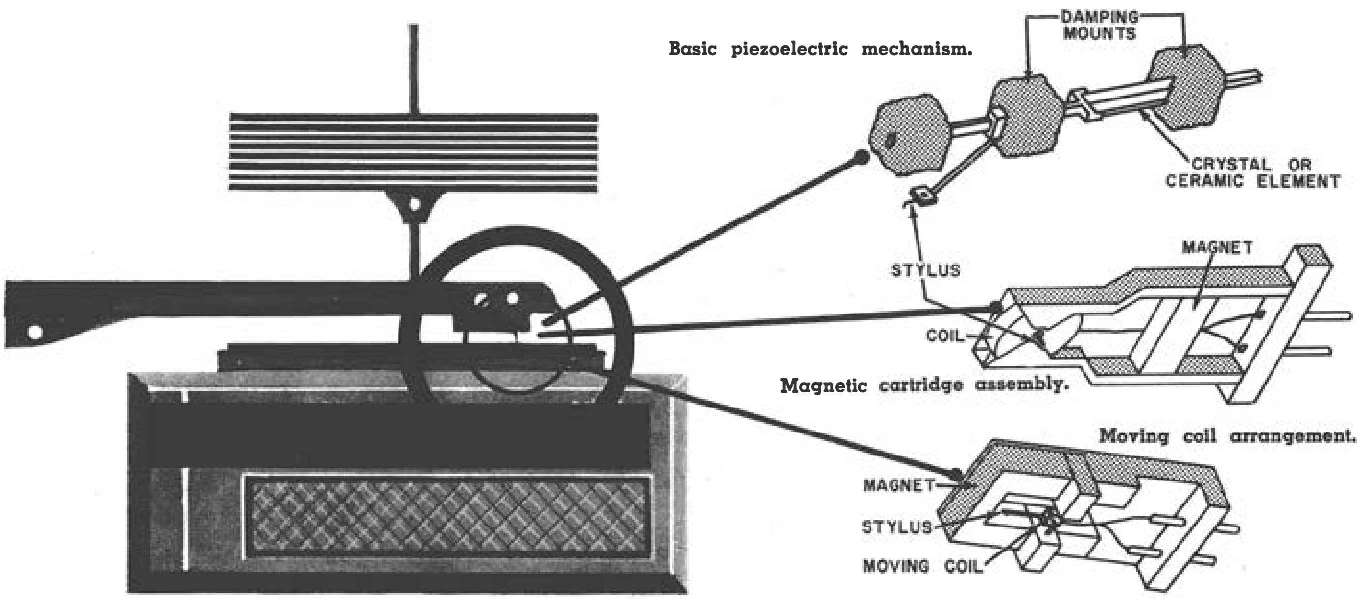
If a booster is used, it may be built as part of the preamplifier proper or on a separate chassis. The shielded leads from the booster to the main amplifiers or control center should be kept short in order to prevent high-frequency attenuation by the cable since the output impedance of the booster is considerably higher than that of the preamplifier.

The primary source of noise in the preamplifier is due to thermal noise in the loading resistors. Therefore, be sure to use low-noise deposited carbon types for R_1, R_2, R_9 and R_{10} as well as for the collector resistors R_3 and R_{11}.

A double-shielded cable is recommended for connecting the cartridge to the preamp. However, double-shielded cable may be difficult to obtain. A simple solution is to use conventional shielded cable, sliding copper braiding over the cable to form the second, grounding shield. Cable capacitances of up to 500 $\mu\mu$f. can be tolerated without any affect on the performance.

No adjustments are necessary in the preamplifier, but the perfectionist may wish to trim the RIAA equalization to follow the RIAA curve exactly. This adjustment is not essential because the RIAA curve will be followed within a decibel with the nominal components listed. Confidentially, the author trimmed the values in his unit just to make *sure.*

If you wish to adjust the preamp to follow the RIAA curve exactly, proceed as follows: Set an input generator to 1000 cps and adjust its amplitude so that the preamp output is .1 volt. Set the generator to 500 cycles and trim C_4 (and C_{10}) so that the output rises 3 db above the 1000-cps level (1.41 volt). Set the generator to 2120 cycles and trim C_3 (and C_9) so that the output drops 3 db below the 1000-cps level (.707 volt). The preamplifier is now adjusted to the RIAA characteristic.



LOADING THE PHONO CARTRIDGE

By **HERMAN BURSTEIN**

Get the best performance from your hi-fi pickup by using the proper cartridge loading circuits.

CORRECT loading of the phono cartridge pays dividends in terms of the smoothest and widest frequency response that the cartridge permits and, often, reduced distortion as well. This is true for both magnetic and piezoelectric (crystal and ceramic) pickups—the two most commonly employed types. Load requirements differ widely between magnetic and piezoelectric units. In addition, there are considerable differences among magnetics and among piezoelectrics; merely connecting the cable from the cartridge to a jack designated for a magnetic or piezoelectric pickup doesn't automatically guarantee that all will be well.

The load across a cartridge is both resistive and capacitive. The resistive load consists of a resistor (sometimes a potentiometer or fixed voltage divider) across the input jack or the grid resistor of the input stage, or a combination of the two. The capacitive load includes cable capacitance and input capacitance of the first stage. The latter consists mainly of grid-plate capacitance, which can be quite large in the case of triodes, due to the Miller effect; there is also grid-cathode and stray wiring capacitance. Altogether, it is not unusual for input capacitance to be 50 μμfd. or more in the case of triodes.

In determining load requirements, the inductance, winding capacitance, and d.c. resistance of magnetic cartridges and the capacitance of piezoelectric cartridges have to be taken into account. The following discussion will review the basic principles of loading magnetic and piezoelectric cartridges, taking all the above-mentioned factors into consideration and listing the impedances of a number of popular cartridges and the loads recommended by their manufacturers.

Magnetic Cartridges

Fig. 1 shows the principal circuit impedances when a magnetic pickup is used. The winding contains inductance and d.c. resistance in series, shunted by the cartridge's winding capacitance, the load resistance, and the load (cable and input) capacitance in parallel.

Omitting the resistances, the inductance and total shunt capacitance form a series-resonant circuit, as shown in Fig. 2. Output, across C, is maximum at approximately the resonant frequency, $f = 1/(2\pi\sqrt{LC})$. Therefore response is not smooth but exhibits a peak at f, as shown in Fig. 3. If L and C are relatively small, electrical resonance occurs far outside the audio range and is not a problem.

However, high-impedance cartridges, such as the variable-reluctance type, have relatively large inductance, which together with typical circuit capacitance brings f dangerously close to or within the audio range. To illustrate, assume L is .6 henry, which is fairly representative, and that C is 400 μμfd., in large part due to a long length of high-capacitance cable. Therefore f is slightly above 10,000 cycles, producing an audible peak.

One purpose of load resistor R_L is to damp the peak. The smaller R_L, the smaller is the resonant peak. But most things have a price and in the present case what happens is that L and the circuit resistances form a low-pass filter, as shown in Fig. 4. The circuit resistance $(R_L + R_w)$ must be large enough to prevent a significant dip in response below 15,000 cycles or so. For example, if L is .6 henry and we select 20,000 cycles as the point where we are willing to let response drop 3 db due to low-pass filter action, then $R_L + R_w = 2\pi fL = 75{,}000$ ohms. Assume that R_w is 1000 ohms. Thus, R_L should be 74,000 ohms. The nearest standard value, 75,000 ohms, is adequate. R_L is not critical and any value within ±10% of the calculated one is generally suitable.

Across the magnetic phono input jack of many control amplifiers there is a load resistor of 47,000 ohms or thereabouts. However, as shown by the example just cited, this can be far too low for some cartridges and, for others, it may be too high. It is important to make whatever changes are required to provide a correct load for the cartridge employed.

Table 1 shows, to the extent that the author has been able to obtain the data, the inductance, d.c. resistance, winding capacitance, recommended load resistance, and permissible load capacitance as given by the manufacturers of a number of popular magnetic pickups. Permissible shunt capacitance (winding, cable, and input capacitance) can be calculated by the formula $C = 1/(4\pi^2 f^2 L)$. L is the cartridge inductance, while a value of 12,000 to 15,000 cycles may be used for f. For ease of calculation, using 15,000 cycles as f, the formula may be put into the form: $C = 113/L$ with C in μμfd. For example, if L is .6 henry, C is about 190 μμfd.

The reader may wonder why, as long as the resonant peak is damped by R_L, it is necessary to keep the resonant

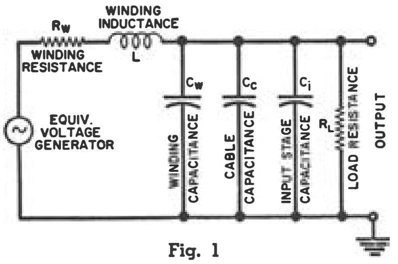


Fig. 1

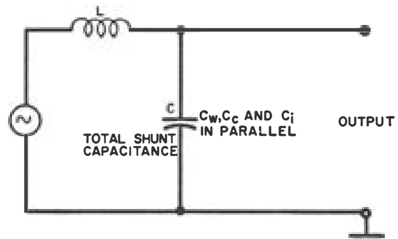


Fig. 2

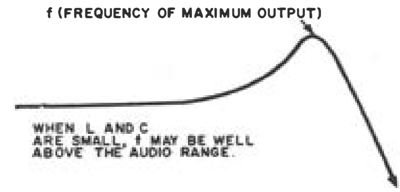


Fig. 3

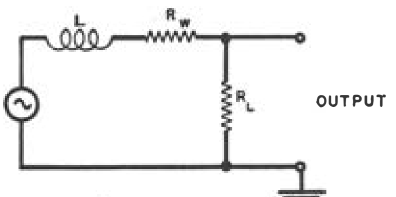


Fig. 4

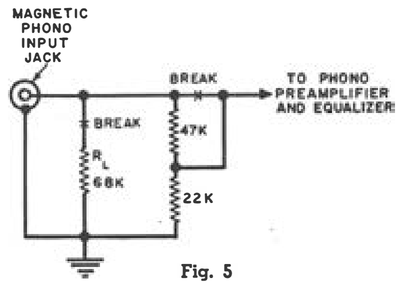


Fig. 5

Fig. 1. Circuit parameters that are associated with a magnetic phono cartridge.

Fig. 2. Resonant circuit produced by winding inductance and shunt capacitance.

Fig. 3. Peak in output due to circuit resonance before adding damping resistor.

Fig. 4. Low-pass filter action produced by inductance and resistances in Fig. 1.

Fig. 5. Converting load resistor to a voltage divider to avoid preamp overload.

peak well up in the audio range or just above it. The answer is that response drops very sharply above resonance, as shown in Fig. 3.

Phono pre-equalization cuts the lows and boosts the highs, requiring the opposite equalization in playback, namely bass boost and treble droop. In a few preamplifiers it has been the practice to supply only bass boost, with treble cut obtained through a suitably small load resistor (for example, the *G-E* Model A1-203); as explained in connection with Fig. 4, winding inductance and circuit resistance—mainly the load resistor R_L—produce treble attenuation.

RIAA equalization, almost universally used today, requires that playback response be 3 db down at 2122 cycles and continue to drop at a rate approaching 6 db per octave. The required circuit resistance is given by $R_L + R_w = 2\pi fL = 6.3 \times 2122 \times L$. To illustrate, if L is .6 henry, then the required circuit resistance is about 8000 ohms. Assume the d.c. resistance of the coil is 1000 ohms. Then R_L should be 7000 ohms. In this case the nearest standard value, 6800 ohms, would be used.

This method of achieving treble cut, prior to the preamplifier, has both advantages and disadvantages. Use of a very small load resistor almost completely damps out resonance. The reduced signal presented to the first stage tends to decrease distortion. A considerable amount of load capacitance can be tolerated without ill effect. On the other hand, signal-to-noise ratio of the playback system is smaller if treble cut takes place before rather than after the input stage; treble cut after the input stage simultaneously reduces noise of this stage.

The problem of electrical resonance ordinarily does not concern the moving-coil type of cartridge, which typically has but a few millihenrys inductance, so that practical values of circuit capacitance place resonance far above the audio range. The load resistor can range in value from a few hundred ohms up to high-impedance loads. The low inductance of the cartridge makes it impractical to obtain treble cut through a suitable load resistor, which would have to be extremely small. Moreover, the relatively low output characteristic of a moving-coil pickup (in exchange for high-quality performance) makes it all the more desirable to obtain reduction of input stage noise concomitant with treble attenuation following this stage.

Some magnetic pickups have quite a large signal output—as much as 100 millivolts on peaks. This might overload the first stage, hence it may be desirable to attenuate the signal from the cartridge. This can easily be done by converting the load resistor to a voltage divider, as illustrated in Fig. 5. Some control amplifiers have an extra input jack intended for high-level magnetic pickups, with a voltage divider across the jack. However, it may be necessary to change these values in order to make their total resistance equal to the load requirement of the cartridge.

Piezoelectric Pickups

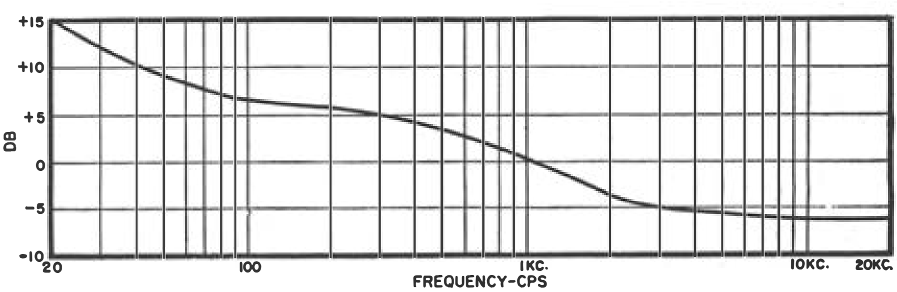
Whereas the output of a magnetic cartridge is proportional to groove velocity, the output of a piezoelectric cartridge is proportional to groove amplitude. As a result of RIAA pre-equalization, groove amplitude varies with frequency in the manner shown by Fig. 6; there is a good deal of bass boost and a moderate amount of treble cut on

CARTRIDGE	INDUCTANCE	D.C. RES.	WINDING CAP.	RECOMMENDED LOAD RES.	MAXIMUM LOAD CAP. (Note A)
Audiogersh MST-1 (Note B)	320 mhy.	1400 ohms	Note C	47,000 ohms	200 μμfd.
Audiogersh MST-2	Note C	Note C	Note C	100,000 ohms	200 μμfd.
Electro-Sonic C-60	1 mhy.	40 ohms	Unmeasurable	100 to 100,000 ohms	Very large (Note D)
Electro-Sonic P-60	1 mhy.	40 ohms	Unmeasurable	100 to 100,000 ohms	Very large (Note D)
G-E VR-II (Note E)	520 mhy.	600 ohms	about 50 μμfd.	100,000 ohms	300 μμfd.
G-E VR-II (Note F)	250 mhy.	400 ohms	Note C	100,000 ohms	300 μμfd.
Grado F28D	Very low	600 ohms	Note C	5000 ohms & up	Note J
Norelco AG3121	600 mhy.	1200 ohms	70 μμfd. (Note G)	68,000 ohms	250 μμfd.
Shure "Studio Dynetic"	130 mhy.	180 ohms	40 μμfd. (Note H)	10,000 ohms	1500 μμfd.
Shure "Professional Dynetic"	350 mhy.	440 ohms	30 μμfd. (Note I)	27,000 ohms	Note C
Tannoy "Mark II"	320 mhy.	1100 ohms	230 μμfd.	100,000 ohms	150 μμfd.

NOTES: A = cable capacitance and input stage capacitance; B = model with diamond stylus; C = information not supplied by manufacturer; D = the manufacturer states, "The cartridge will be unaffected by any cable or tube capacitances encountered"; E = high-impedance model; F = low-impedance model; G = calculated on basis of resonant frequency at 25,000 cycles, per manufacturer; H = calculated on basis of resonant frequency at 70,000 cycles, per manufacturer; I = calculated on basis of resonant frequency at 50,000 cycles, per manufacturer; J = cables in excess of 10 feet may be used.
For information on cartridges not covered, contact the manufacturers direct.

Table 1. Characteristics and loading for some representative magnetic cartridges.

Fig. 6. Variation of the groove amplitude with frequency due to RIAA recording equalization, assuming there is a constant signal input at all the frequencies depicted.



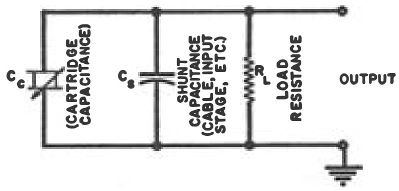


Fig. 7. Circuit parameters that are associated with a piezoelectric cartridge.

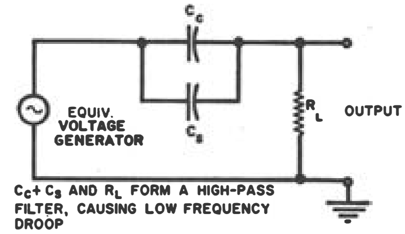


Fig. 8. Equivalent circuit of Fig. 7 with respect to low-frequency response.

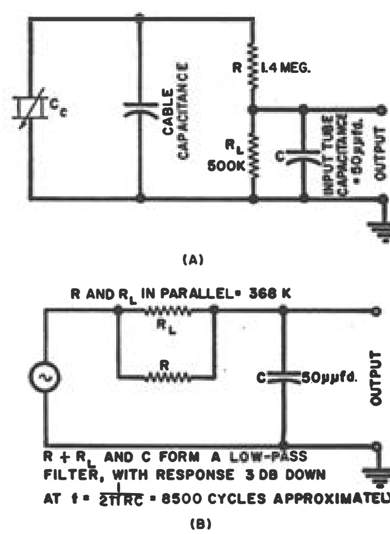


Fig. 10. (A) Effect of using a voltage divider as load resistor for piezoelectric cartridge is shown here. (B) Equivalent circuit of Fig. 10A at high frequencies.

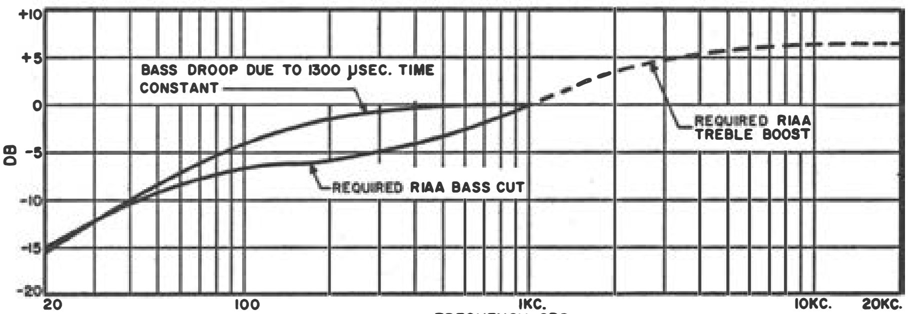


Fig. 9. Comparison of RIAA bass cut required when using amplitude-responsive pickup with bass droop produced by circuit impedances having time constant of 1300 μsec.

the record. Thus, when a piezoelectric cartridge is used for playback, bass *cut* and treble *boost* are required, which is just the opposite of the requirements for a magnetic cartridge.

Fig. 7 shows the basic circuit elements when a piezoelectric cartridge is used. Fig. 8 is the equivalent circuit and it can readily be seen that the circuit capacitances, effectively in parallel as far as frequency response is concerned, form a high-pass filter in conjunction with the load resistor. In other words, these elements produce bass cut. To achieve bass cut that corresponds fairly accurately to RIAA requirements for an amplitude-responsive cartridge, the time-constant of the total circuit capacitance and the load resistor should be between approximately 1000 and 1300 microseconds. This means a bass droop characteristic with a turnover frequency (3 db decline) between 160 and 120 cycles and declining thereafter with decreasing frequency at a rate approaching 6 db per octave. Fig. 9 compares the required RIAA bass droop with that achieved by circuit elements having a time-constant of 1300 microseconds (120-cycle turnover frequency).

The principal circuit capacitance is usually that of the cartridge, varying from about 400 μμfd. to as much as 2000 μμfd. The other circuit capacitances, chiefly that of the cable and the input tube, have a total value which typically ranges from 150 to 250 μμfd., although a long run of high capacitance cable can raise this figure appreciably.

To illustrate the method of calculating the required load resistance to produce the correct amount of bass droop, assume that the pickup has 500 μμfd. capacitance and that cable and input tube capacitance totals 200 μμfd. To obtain a time-constant of, say, 1300 microseconds with circuit capacitance of 700 μμfd., the load resistor has to be about 1.9 megohm. The nearest standard values of 1.8 or 2 megohms will work satisfactorily.

The signal output of a piezoelectric cartridge may, in some cases, reach as much as 1 or 2 volts on peaks, indicating that these pickups are meant to be connected to high-level inputs. But the typical input impedance at the high-level jack of a control amplifier is 500,000 ohms. This is too low for most piezoelectric cartridges, that is, the time-constant of the circuit capacitances and load resistor will cause too much bass cut.

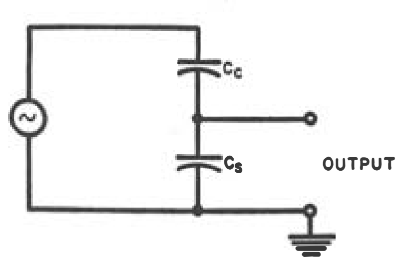
The remedy is simple: either (1) increase the input impedance to the required value, perhaps by changing the grid resistor of the input tube, if its characteristics permit, or (2) place a capacitor of suitable value across the cartridge to increase the circuit capacitance.

The second method is simpler since the capacitor can be mounted directly across the cartridge terminals, making it unnecessary to go into the control amplifier. To illustrate, if the load resistance is 500,000 ohms, then 2600 μμfd. of circuit capacitance is required for a time-constant of 1300 microseconds. Assuming the capacitance of the cartridge, cable, and input stage total 700 μμfd., then another 1900 μμfd. of capacitance is required. A capacitor of .002 μfd. across the cartridge would be close enough in value to achieve the objective of correct bass attenuation.

If the volume control of the control amplifier follows rather than precedes the input stage for high-level sources, this stage may be overloaded by the signal from a high-output cartridge. It might seem that the simplest way to attenuate the signal would be through use of a voltage divider as the load resistance (see Fig. 5). Generally, however, this is a poor idea because it may well entail substantial high-frequency loss. Fig. 10 shows why. Fig. 10A represents a voltage divider having a total resistance of about 2 megohms and with the values chosen to produce about 12 db attenuation. The input capacitance of the first stage is assumed to be 50 μμfd., a plausible value. Fig. 10B is the equivalent circuit for high-frequency response. The load resistors are effectively in parallel and this parallel value forms a low-pass filter in conjunction with the input capacitance. For the values shown, namely, effective parallel resistance of 368,000 ohms and 50 μμfd. capacitance, high-frequency response is 3 db down at about 8500 cycles, declining thereafter at a rate approaching 6 db per octave.

Thus it is better to attenuate the signal by means of a shunt capacitor of suitable value connected across the cartridge. Fig. 11 shows how the cartridge capacitance and the other circuit capacitances effectively form a voltage divider. The bottom leg of the voltage divider consists of the parallel value of all the circuit capacitances other than the cartridge. If the capacitance of the bottom leg is increased, its reactance decreases and the output across this

Fig. 11. Shown below is the equivalent circuit of Fig. 5 with respect to the voltage divider action produced by means of the shunt capacitance of the circuit.



reactance decreases correspondingly. If the total circuit capacitance is increased by addition of a shunt capacitor, then, as previously explained, the load resistance must be correspondingly decreased to maintain the same time-constant as before. It may well work out that if the load resistance is 500,000 ohms, the additional shunt capacitance required may also provide a satisfactory amount of signal attenuation.

As pointed out earlier, treble boost is required when an amplitude-responsive cartridge is used (see Fig. 9). Ordinarily this is provided by the piezoelectric cartridge through mechanical resonance, controlled in amplitude and range by damping and other design techniques to approximate the RIAA equalization requirement.

It may be desired to convert the piezoelectric cartridge into a velocity-responsive device so that it may be fed into an input jack intended for a magnetic pickup. In most control amplifiers this has the advantage of allowing the user a choice of playback equalization characteristics, some intended for 78 rpm records and others for 33⅓ rpm discs made prior to the adoption of the RIAA curve.

There are two simple means for such conversion: (1) a small capacitor can be placed in series with the cartridge as shown in Fig. 12. This produces high-pass filter action so that output rises with frequency as in the manner of a magnetic cartridge, or (2) a small load resistor can be placed across the cartridge, as in Fig. 13. This, too, results in rising output with frequency.

Neither the series capacitor nor the shunt resistor should be so small as to produce high-pass action throughout the treble range. As already mentioned, the cartridge has built-in treble boost due to mechanical resonance. Therefore, the rise in output should level off at an appropriate point in the treble range, *viz.,* at about 5000 cycles.

To illustrate, assume that the load resistor in Fig. 12 is 500,000 ohms. In order to have a turnover frequency of approximately 5000 cycles (rise in output within 3 db of maximum), we find the necessary series capacitance by means of the formula $C = 1/(2\pi fR) =$ 64 $\mu\mu$fd. (approximately). The nearest standard value would be satisfactory. In the case of Fig. 13, assume that total circuit capacitance is 700 $\mu\mu$fd. We find the necessary load resistance by means of the formula $R = 1/(2\pi fC)$. Using the figure 5000 cycles as f, R turns out to be about 46,000 ohms; a 47,000 ohm resistor will do.

If signal output is excessive after the cartridge is converted to a velocity-responsive device, signal attenuation can easily be achieved by using a voltage divider as the load resistance. As long as the total load resistance is less than about 250,000 ohms, it is not likely that high-frequency loss, such as discussed in connection with Fig. 10, will occur.

Manufacturers of piezoelectric cartridges sometimes recommend a more complex network than a simple series capacitor or shunt resistor to make the pickup velocity-responsive. The purpose of the network is to obtain somewhat flatter response.

Table 2 shows the capacitance, recommended load resistor, and permissible shunt capacitance with this load resistor as given by the manufacturers of a number of popular piezoelectric cartridges. The table also indicates the means, if any, suggested by the manufacturer for making the cartridge velocity-responsive.

For those few special cartridges, such as the FM type, which do not fall in either of the categories discussed, the specific recommendations outlined by the manufacturer must be followed. Also, for information on magnetic and piezoelectric cartridges not covered in Tables 1 and 2, the cartridge manufacturer should be contacted for the correct information.

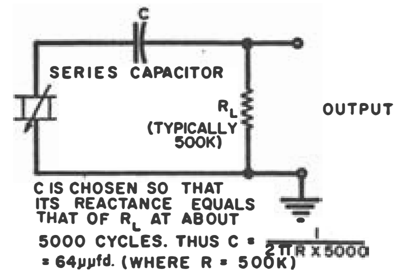
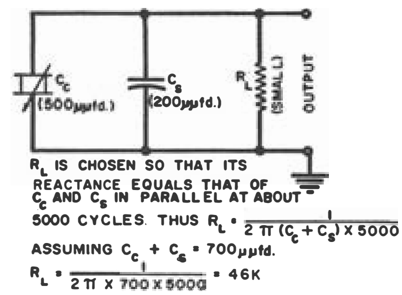


Fig. 12. Small series capacitor converts crystal pickup to velocity-responsive unit.

Fig. 13. Use of small load resistor makes crystal cartridge velocity-responsive unit.



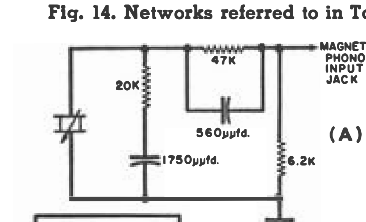
CARTRIDGE	CAPACITANCE	RECOMMENDED LOAD RES.	MAXIMUM LOAD CAP. (Note A)	NETWORK FOR CONVERSION TO VELOCITY DEVICE
Astatic 81-TB	400 $\mu\mu$fd.	2 meg. (Note B)	100 $\mu\mu$fd. (Note B)	Fig. 14A
Astatic 89-TB	600 $\mu\mu$fd.	2 meg. (Note B)	100 $\mu\mu$fd. (Note B)	Fig. 14A
Electro-Voice 8D	300 $\mu\mu$fd.	3 meg.	100 $\mu\mu$fd.	Fig. 14B
Ronette TX-88	1500-1800 $\mu\mu$fd.	500,000 ohms	Note C	Fig. 14C
Shure ML-44	525 $\mu\mu$fd.	Fig. 14D	Note C	Fig. 14E
Sonotone 3T-S	490 $\mu\mu$fd.	2.2 meg. (Note D)	100 $\mu\mu$fd.	Fig. 14G
Webster MC-1	510 $\mu\mu$fd.	.51 to 3.3 meg.	100 $\mu\mu$fd.	Note C
Zenith	500 $\mu\mu$fd.	1 meg.	100 $\mu\mu$fd.	Note E

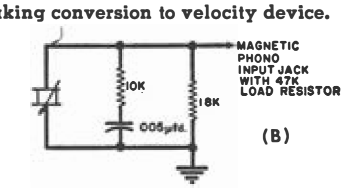
NOTES: A = cable capacitance and input stage capacitance. This is the maximum load capacitance for the recommended load resistance. Higher load capacitances are permitted with smaller load resistances, as discussed in text; B = the manufacturer states, "Load resistance, in megohms, multiplied by total capacitance (cartridge, cable, etc.) should equal approximately 1000"; C = information not supplied by manufacturer; D = see Fig. 14F for alternate load network recommended by the manufacturer for maximum flatness of response; E = the manufacturer states, "We do not recommend the conversion of this pickup into a velocity device."

For information on cartridges not covered, contact the manufacturers direct.

Table 2. Characteristics and loading for representative piezoelectric cartridges.

Fig. 14. Networks referred to in Table 2 for making conversion to velocity device.

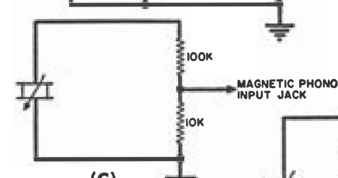


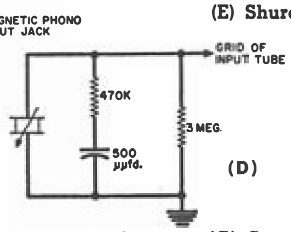


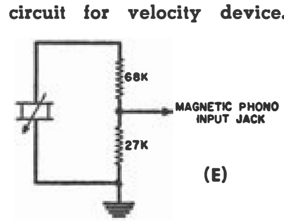
(A) Network given for Astatic pickup.

(B) Network suggested by Electro-Voice.

(E) Shure circuit for velocity device.



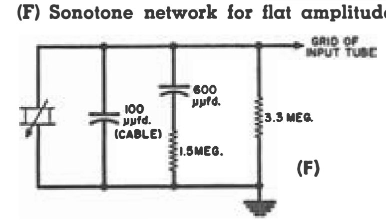




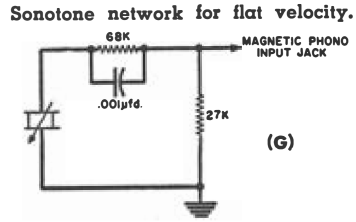
(C) Ronette circuit.

(D) Shure network for flat amplitude output.

(F) Sonotone network for flat amplitude.



(G) Sonotone network for flat velocity.



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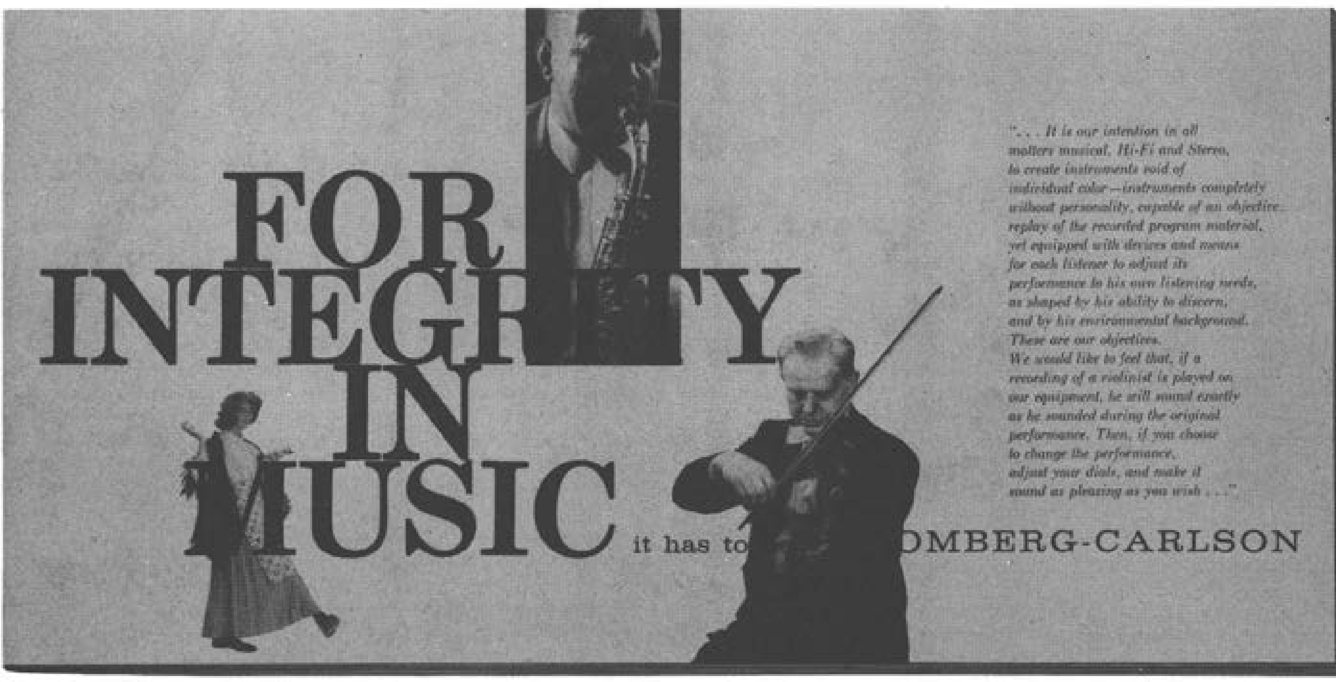

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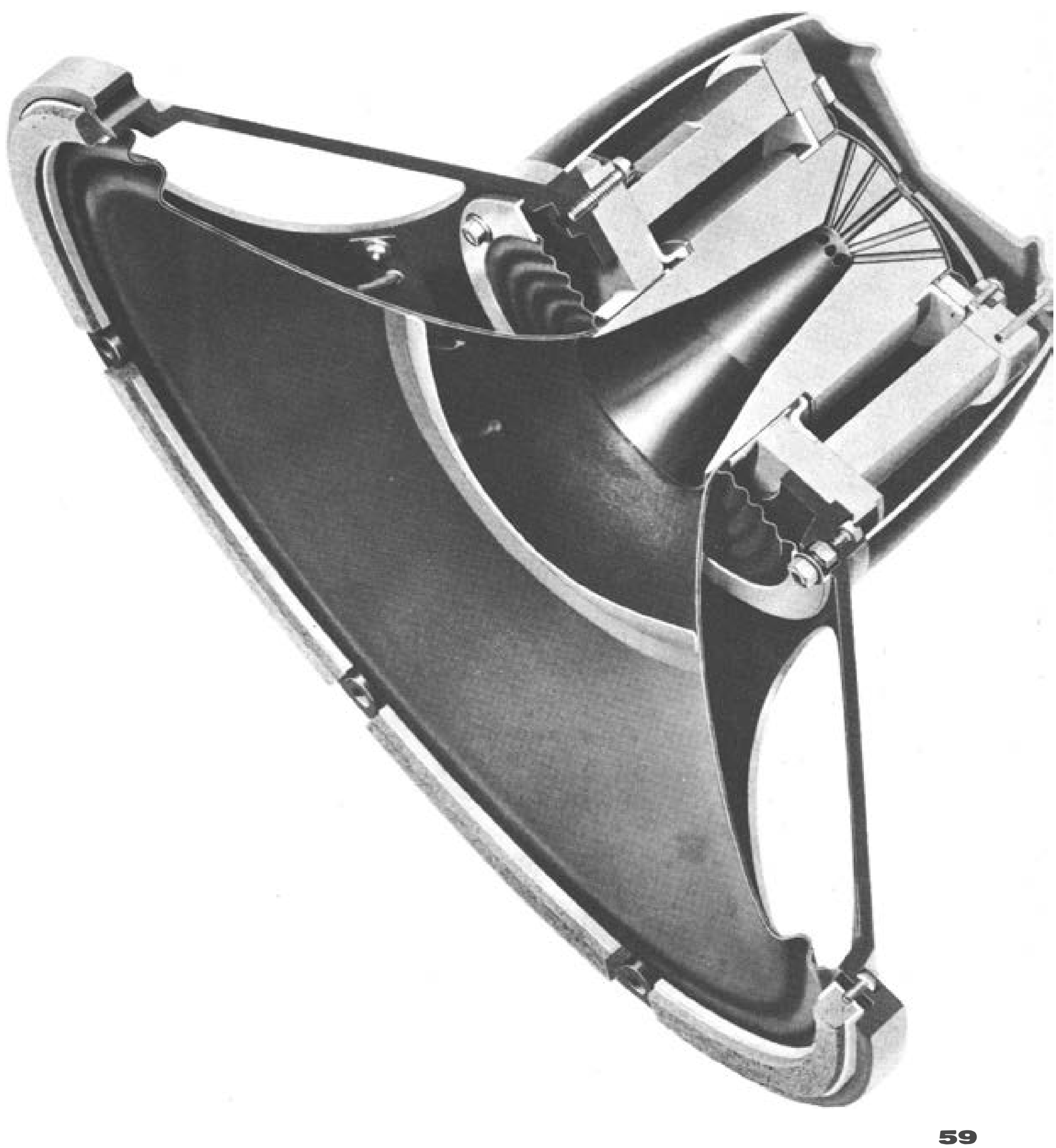
Section III

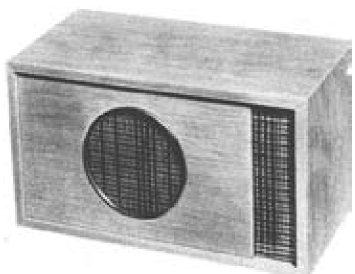
Loudspeakers and Enclosures



Hartley 217-Duo

University "Stereoflex II"

Ampex Model A-423



Sonotone "Caprice"

Bozak Model B-304

General Electric A1-406

Heathkit SE-1 and SC-1's

Loudspeakers for Stereo

By **NORMAN H. CROWHURST**

PART 1 **What speaker system should you use for stereo? This article will help you to make a choice.**

STEREO is big news these days and naturally everyone wants "a piece of the cake." The gimmick artists are well in evidence. This is a pity, because stereo can get quite confusing enough if we stick to the strictly legit end.

"Hear your hi-fi *in three dimensions!*" is the substance of some stereo promotion, which carries the implication that, until now, our poor little ears must have been struggling along in only two—or that we have only been using one ear! The fact is that, for two-eared people, listening has been a stereophonic experience from birth. So what do we really expect stereophonic sound to add to high fidelity?

This is the first thing we should get straight. It will help us a lot in understanding how to pick and use loudspeakers to get the best from stereo program material. Single-channel high fidelity, nowadays called monaural (which admittedly is not a good name, because it means "one-eared") or monophonic, has the limitation that an original performance occupying four dimensions (waves in three-dimensional space and time) must be compressed into two (*magnitude* of fluctuation with *time*) for recording or transmission. This must lose some of its identity, although the reproducer puts it back in four dimensions again. But it is impossible, from the limited "data" that can be conveyed in a two-dimensional channel, to reconstitute the original with four-dimensional precision.

Our hearing faculty has acquired the capacity to give us a quite accurate impression of the four-dimensional events going on around us from an analysis of two "two-dimensional channels," one received by each ear. But analysis and synthesis are different things.

Our hearing faculty can give a wonderful sound picture of the world around us from this analysis; but it would be quite impossible to use this two-channel-transmitted sound picture to recreate the world of sound: a dummy head with the most complex electronics in the world could not *produce* the noise of an aircraft 5000 feet up! Nor can two loudspeakers project an orchestra. This idealized concept of stereo is certainly a basic fallacy.

The viewpoint that is more successful as an approach to stereo is that conveying the total sound on two channels gives us twice the potential that one does for achieving a realistic illusion. In particular, this potential lies specifically in the improvement of perspective.

So we should visualize stereo as a system in which, basically, we have two channels instead of the one used in mono. This may seem obvious, but it is the only thing that is *basic* in stereo. Beyond these two channels, the success of the illusion depends on what is done at their two ends: the microphone and recording technique at the input; and the playback and loudspeaker arrangement at the output.

Each of these variables provides the possibility for a wide variety of combinations. Microphones with at least three variations in directivity can be used in any number and placement, and their outputs combined in different

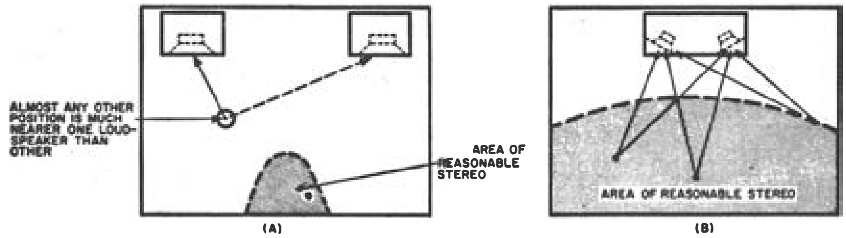


Fig. 1. Spaced speakers in a small room (A) may restrict the area of acceptable stereo. The composite system in a single enclosure (B) may do a better job here.

ways to produce the final two stereo channels. There is an almost similar range of possibility for variation in the use of loudspeakers.

But we've said enough to show that stereo is not the simple thing some theorists have suggested. Instead of proceeding further on how complicated it can get, let's deal with some specific questions.

1. *Some recommend a stereo system with two loudspeaker systems in a single enclosure, while others insist stereo can only be obtained with two speakers spaced apart along a wall. Which is best?*

This depends. Each can be "best" in circumstances suited to it. Probably most important of these is room size. In a small room it is practically impossible to find more than a very small listening area where one is not much nearer to one speaker than the other, if spaced units are used. (Fig. 1A). By using a single enclosure system, all parts of the room are at a commensurate distance from the two loudspeakers (Fig. 1B) and the stereo pattern is created in any location by the transverse component of the sound wave, caused by the *difference* in radiation from the two units (Fig. 3).

But in larger rooms the relative merit is almost exactly reversed. The transverse component of the wave only holds a reasonable strength for a very small distance from the composite loudspeaker system, so the successful listening area will be confined to a narrow line down the middle and a slight enlargement right at the front (Fig. 2A). On the other hand, the larger room dimensions enable spaced speakers to "push out" the stereo effect further and only comparatively small areas of the room are now uncomfortably close to one speaker (Fig. 2B).

A secondary factor is the way the program was miked. This may be a deciding factor in medium-sized rooms —say about 15' by 20' or a little larger. The hearing faculty bases its analysis on the time difference of various sound components in the composite wave *reaching each ear*. But to maintain this correct time difference, without exaggerating or reducing it, proper proportion must be kept between time and intensity differences.

If widely spaced mikes are used for pickup, combined with widely spaced loudspeakers for playback, the time difference is going to get exaggerated at some listening locations in the room to the point where an echo, or double image, rather than perspective results. This would be particularly pronounced on material such as strings played *pizzicato*.

On the other hand, if time difference on the channels is small and intensity difference has been accentuated, either by using directional mikes close together but facing different directions or by using electronic mixing of mikes close in to individual instruments, the "projection" of the stereo illusion will be strictly a function of speaker spacing to "create" a time difference at the listener's ears.

Thus, within certain room size limitations, program made with a wide-spaced mike technique should use loudspeakers close together (in one cabinet) while program that used close-together (directional) mikes should be reproduced on widely spaced loudspeakers.

2. *Different, and quite conflicting, statements have been made about the contribution made by various parts of the audio spectrum to the stereo illusion. Just what are the important "stereo" frequencies?*

Unfortunately, a lot of work has been done using continuous steady tones, either fed through headphones or using multiple loudspeakers. From these experiments various deductions have been made as to the dependence of our sense of direction on intensity and phase differences at different frequencies. The deductions conflict because of different measurement techniques.

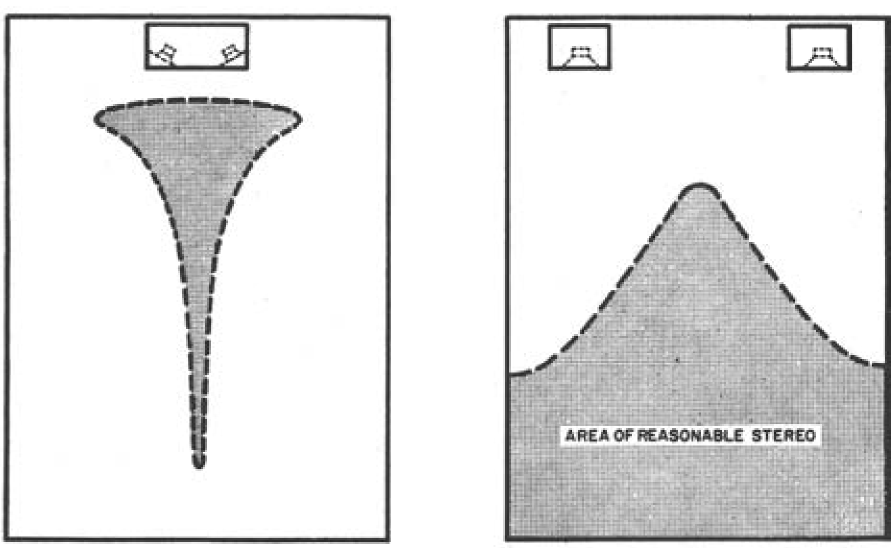


Fig. 2. Composite system (A) may be limited in coverage of larger room. Spaced speakers (B) may do a bettter job here.

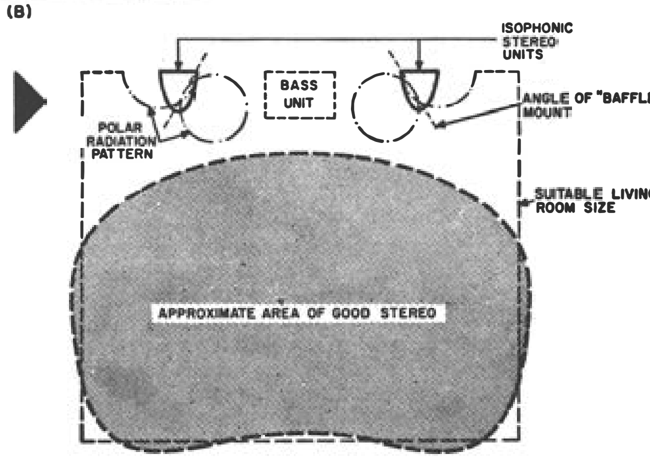
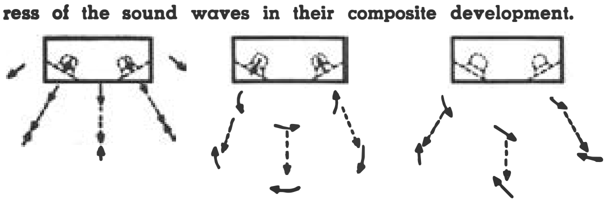


Fig. 4. Coverage claimed for CBS "Isophonic" system.

Fig. 3. Synthesis of radiation with both units in-phase (A), out-of-phase (B), and when radiation represents a source to extreme left (C). Solid arrows are instantaneous motion due to sound wave; broken arrows are used to represent the progress of the sound waves in their composite development.



However, other work shows that these results are not relevant to the stereo illusion perceived on "live" sounds. A particularly effective demonstration of this fact occurred in work with the *Perspecta* system of stereo for theaters. If, for example, the initial beat of a drum is accurately located by the stereo illusion, it is practically impossible to tell that the follow-through "oing" shifts clear across stage, *even when you know it does.*

This is equally true of other types of sound. Only sounds generating transient components continuously, such as speech or a succession of different notes being played, enables a moving source to be followed. A tone containing repetitive sharp transients, such as an aircraft motor, will also make this possible.

The hearing faculty seems to identify the direction of the *composite* transient, rather than of individual *frequencies* it contains. If the time difference is identical for all these component frequencies, then the sense of "integrity," both of position and the sound itself, is improved.

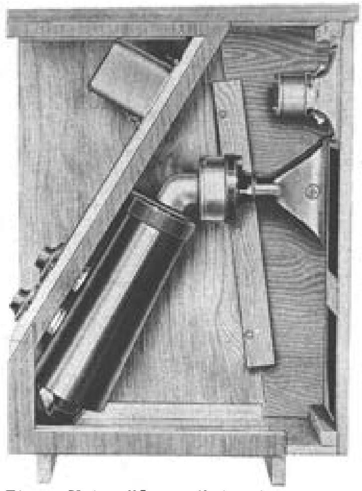
But what does this mean, as applied to reproducers for stereo? The difference in conclusion largely concerns the higher frequencies, from 1000 or 1500 cycles up. It occurs because different methods are used to "generate" the right differences at the ears of the listener.

With the wide-spaced loudspeakers, difference in intensity of radiated sound produces a time difference at the ears, due to the obstacle effect of the head, that depends on frequency. At higher frequencies the difference is likely to be altogether greater than for lower frequencies, invalidating a proper association of the high-frequency components with the lower ones. So with this arrangement, it is not surprising that critical tests show that the very-high-frequency components do not contribute anything to the stereo illusion.

With speakers close together, and employing directional radiation, the higher frequency contribution to the stereo illusion is due to the precedence effect. A speaker directly facing the listener, and having the greater intensity, will localize itself as the source. If the higher frequencies come from the other speaker in greater intensity, however, the source will be delocalized and the hearing faculty will associate, by comparison, these components with the lower frequencies to which they belong. Thus use of a different type of speaker enables the higher frequencies to contribute a useful part to the stereo illusion.

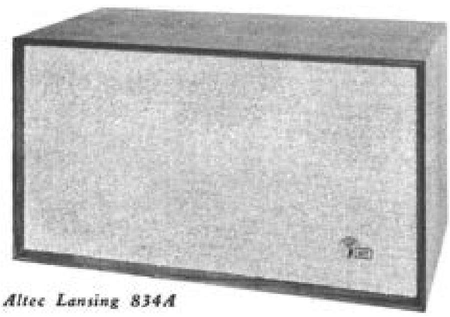
3. *What about the statement that the same loudspeakers used on stereo show an improved frequency response?*

Common sense urges that this cannot be so—how can a loudspeaker "know" whether it is handling mono or stereo? And yet people who have experimented with it swear they can hear the difference.

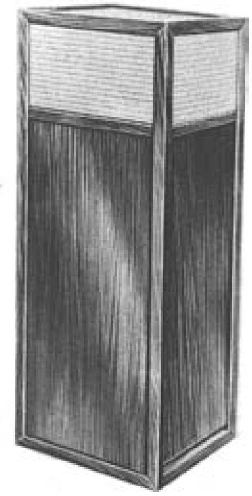


Electro-Voice "Stereon" interior

Knight Model KN-2000



Altec Lansing 834A



EICO Model HFS-2

Extension of apparent range at the low end is relatively simple and there is little argument about it. Two units, even on the same program, will put out more bass than one, other things being equal. The same improvement in this direction naturally shows up on stereo. But this cannot be the reason at the high end.

Did you ever hear any audio components above 4000 or 5000 cycles? You've heard background hiss in this range and you've heard the improved clarity of certain instruments when this range is there, compared with when it is missing. But have you heard those components *by themselves?* This is the answer. You identify presence of these frequencies by the improved clarity they bring to individual instruments you could *hear* without them.

Stereo also does this, by a different means. It improves clarity of individual instruments by giving them separate localization. And because both effects serve the same purpose to the hearing faculty—improved clarity—it takes an unusually well-educated hearing to detect the difference. A definite illusion of improved high-frequency response—as we have become familiar with it—is obtained.

4. *What about the idea of using one speaker for all the bass with separate speakers for treble?*

Some say this can be done—in fact they do it, while others say separation of the lower frequencies into the two channels is definitely necessary to stereo. This question hinges on "lower than what?" What crossover is used? I asked one speaker manufacturer who took the second viewpoint what crossover he used. He had only tried it with one of his regular units, using his regular crossover at a frequency of 1000 cycles.

There's your answer. The people who do it successfully use a crossover of 250 cycles—two octaves lower—or at the highest 400 cycles. This makes a tremendous difference. At 250 cycles the wavelength of sound in air is more than 4 feet; at 1000 cycles it is only just over 1 foot.

At frequencies far below 250 cycles (which in theory may be marginal, but those who use it have checked that it is a satisfactory transition point) the frequency content of both stereo channels is not only sensibly in-phase, there can also be little difference in intensity. So no noticeable "error" is involved by "putting the two together."

"But how about transients?," I have been asked. Now just what is a low-frequency transient? Some visualize a low frequency, below 250 cycles, that starts abruptly. But a low-pass filter, rolling off at 250 cycles, will not allow such an abrupt start. If a 250-cycle tone is keyed on, or started with a blast, this start will contain a range of components *above 250 cycles,* not below it. And the click, or burst, associated with the commencement of the 250-cycle tone, will identify its location quite successfully, even though the body of sound always comes from the center loudspeaker!

At one demonstration of a common-bass system, I went to within about two feet of one of the small speakers (they were over 20 feet apart) before my ears could tell the bass was coming from somewhere else. So it definitely works—at that frequency.

5. *When a center speaker is used for common-bass, with side stereo speakers, can some of both channels also be combined and fed to the center location, to give "center fill"?*

This is a "loaded" question. A paper presented by *CBS Labs* at the fall AES Convention stated that using any such center fill would destroy proper stereo, while the *Stephens* "Stereodot" system actually mixes program from both side channels to feed to the center.

But a closer look shows there is good reason for the apparent contradiction. The *CBS* system—and they only said this in reference to that system—uses what they term an Isophonic loudspeaker for the side locations. This is a small unit mounted on an open baffle, pointing inwards at an angle of 60° (Fig. 4). For esthetic purposes it is housed in an enclosure that makes it look as if it points toward the front.

This system, above 250 cycles, uses the special shaped radiation pattern produced by the open-backed speaker (a figure of 8, of which only part of the front "lobe" is used). Correct sense of "location," in this system, depends on the way the radiation from just the two side "Isophonic" speakers combines. So use of a center fill unit would upset this combination.

"Stereodot," like some systems recommended by other manufacturers, uses small "pressurized" units (with sealed backs) for the side speakers, above the common-bass crossover frequency. This far, the system is conventional, and quite different from the special distribution pattern of the *CBS Labs* system. So it will no more invalidate stereo to use center fill here than it does in any basic two-speaker stereo system, such as the *Klipsch* "Heresy."

I didn't want to mention names in this part of the article, but it seemed unavoidable here. However, my reason for using this information is that it illustrates how using a different type of loudspeaker radiation pattern can completely change things.

6. *Some have asserted that a basic requirement for good stereo is an omnidirectional loudspeaker, while others deliberately use the directional characteristics of loudspeakers to obtain "best stereo effect." Which is best?*

The basis for this difference has already been touched on in answer to the previous questions. The second part of this article will go into more details about actual systems. But the basic facts we can state here.

When the two stereo speaker systems are contained in the same cabinet (or are that close together), the radiation of the middle and upper frequencies *must* be directional for successful stereo. On the other hand, with a certain so-called "ideal" spacing for stereo speakers (which depends on room size, incidentally), best results will be obtained if the units are diffused to the point of an omnidirectional radiation.

But when you don't have a room that suits this "ideal," it needs individual consideration for its particular needs.

Loudspeakers for Stereo

By NORMAN H. CROWHURST

Part 2

Some special speaker setups that represent the various manufacturers' answers to the best way to hear stereo.

A GOOD many loudspeaker manufacturers have put in a lot of work finding out just what "makes stereo." They may have come up with different conclusions (which adds to the confusion), but one can credit them with honest effort, both for their research and in making available products consistent with their "findings."

We discussed the reasons for these differences in Part 1 of this article—so now let us consider some other aspects. Some manufacturers have been quite "purist" and produced only systems that give stereo in its best or "final" form; if you want stereo, according to them, don't settle for any half measures. Others have catered to people who have monophonic hi-fi and want to convert to stereo by adding a channel (as a first step, at least) or for people with practical limitations: no room for the "conventional" set-up or an awkward shaped listening area.

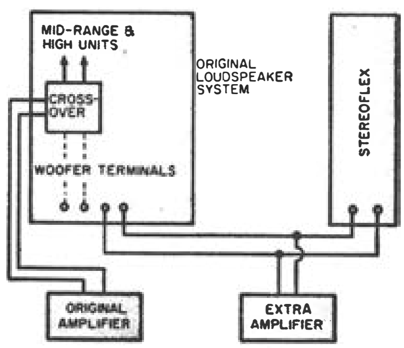
"Add-On Units"

Early in the field of the "add-on" market was *Electro-Voice* with the "Stereon." This is a versatile unit intended to take the mid-range and highs of the second channel (left or right) while the bass of both channels is fed to the bass unit of the original system. A filter is provided to do this, as shown in Fig. 7. *University* has also introduced three add-on "Stereoflex" units. However, with some *University* woofers, the dual voice-coil makes the filter for diverting bass from the second channel to the original woofer unnecessary. One voice-coil of the "common" woofer is fed from each amplifier output by a relatively simple connection (Fig. 5).

This add-on approach is not favored by other manufacturers, however. They feel that in most instances the add-on unit will eventually be abandoned in favor of a more ideal system. While this is probably true, there may also be some places where "one large, one small" fit so well they will remain. It can give quite good stereo presentation, although admittedly, where circumstances allow, better arrangements are possible.

Another approach that can be considered as an "add-on" one is the *Stephens* "Stereodot," although this is really a system approach. However, it can be added on to an existing hi-fi system by including two "Stereodot" side speakers (which are very small) and the control unit. It is an extremely versatile system because the small "Stereodot" units can be mounted almost anywhere. It also provides for "center fill" by feeding a mixture of middle and high frequencies to the original wide-range loudspeaker, as well as using it for the bass of both channels (Fig. 8).

Fig. 5. This is the connection possible with the University "Stereoflex" as employed along with double-wound woofer.



A very versatile system in this category is one used in several *Columbia* consoles. This is not so much an add-on deal as it is a unit with which you can do many things. Primarily designed as the master unit of the "Isophonic" system (mentioned in Part 1) when used this way the console speaker handles common bass only, with the left and right mid-range and highs (above 250 cps) being fed to the small "Isophonic" units. The difference between this arrangement and the *Stephens* system was explained in the previous article. Additionally, a jack-plug arrangement permits the same console to be used with another full-range speaker system, feeding one channel (left or right) to the external speaker and one to the internal speaker. The possible combinations are shown in Fig. 9.

Of course, many other systems can be built on the add-on principle, merely by buying another unit similar to the one you already have and installing the additional electronics somewhere.

It would be impossible to describe here the variety of ways in which this can be done. But we should warn against buying a second *very large multi-unit system.* Not only will the distaff side probably object to your hobby occupying too much of the living room, you will not get the best stereo—in fact you may not get stereo, period! Speakers for stereo *must* give an impression of point-source radiation, if you plan to use two alike, spaced apart in the conventional manner. If you already have one of these large "superdupers," I have two suggestions: either buy a complete separate stereo system and keep the original for mono only or else buy one of the add-on systems, utilizing your single system for part of the stereo—but not for just one channel.

"Unitized" Approach

So much for the add-on approach. Next we turn to the "unitized" approach—putting a complete stereo loudspeaker system for both channels into one cabinet. Several people have done this—in slightly different ways.

Most of these are essentially an approach for the small-to-medium sized room. Some angle the speakers outward for direct radiation (Fig. 6A); some point them out opposite ends with movable reflectors (Fig. 6B); and at least one (*University* "Trimensional") does this with the reflectors in front of the units (Fig. 6C) so as to further utilize wall reflections. Some of them use common and some separate woofers. The *University* version uses a common woofer with twin voice-coils and utilizes radiation between the cabinet back and room wall to improve the low end.

Each of these arrangements uses outward-facing units for the mid-range and high frequencies. Used on stereo material, they each can project sound that appears wider than the piece of furniture from which it actually comes. Choice should be governed by the acoustics of the room in which you will install it, bearing in mind that too little reflection can sound "dead" while too much results in confusion. The direct radiator, without reflectors, will perform best in recreation-type rooms while a type provided with doors that bounce sound off the wall deliberately are better in a room that is "well upholstered."

Quite another type of single-piece-of-furniture entry is the *Ranger-Lansing* "Paragon" (Fig. 10) and its junior version, the "Metregon." These cross-fire the sound into a curved reflecting surface, the object of which is to even out the mean path distance from each

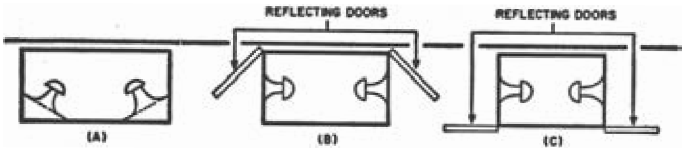


Fig. 6. Three basic constructions used for unitized cabinets.

unit to the listener in various parts of the room, thus spreading the area of acceptable stereo. The reflector alters the apparent position of the two units according to where you sit, so as to optimize stereo in different positions.

This optimizing of the presentation in different positions in the room should not be confused with providing "center fill." Actually the latter is best done by using better microphone techniques in recording. Where this has not been done, a center loudspeaker may help to a limited extent. At the same time, this hole-in-the-middle effect can be more noticeable with some types of loudspeaker than others. The horn type produces an effectively large area sound source and is more prone to exhibit the effect just mentioned than some other types.

For this reason, Paul Klipsch, who advocates horn-type loudspeakers at opposite ends of the longer wall of a room (Fig. 11) developed his "Heresy" for the middle position, together with a simple phantom circuit for connecting it to virtually any pair of stereo amplifiers, so it receives a matrixed signal. This is also the philosophy behind the remixing of middle for the center speaker in the "Stereodot" system.

But optimizing stereo, so its effect can be heard in positions other than center is another thing. This is what various approaches try to do in different ways. The deviant sources of Fig. 6 do it by changing the type of sound distribution from each unit received in different parts of the room.

The cross-fire-with-reflector system on *Ranger-Lansing* does it by shifting the apparent loudspeaker unit positions according to where you sit. The "Isophonic" system of *Columbia* utilizes the radiation pattern in yet another way, to modify receiver intensity from each unit according to where you sit. Each of these three methods works, but produces *different* results. Which is best?

While, as we have said, this may vary with individual rooms in which they are tried, the difference is also subject to individual hearing faculties and experience and although I may not be very helpful in saying this, the only way to know which suits you best is to conduct some careful listening tests of your own.

Stereo Bass

In going over different systems—or for that matter, individual speakers too —you will find a further area of conflict lies in the *kind* of unit used to provide bass response. Paul Klipsch won't hear of anything but a corner horn, except for the center fill. Other manufacturers maintain — as almost anyone would have until a year or so ago—that a speaker must be big to get good bass. Edgar Villchur (of *Acoustic Research*) says that there is merit in a low-efficiency and small bass system.

So what kind of bass is good for stereo? As was explained in Part 1, pure bass is not materially stereophonic in any ordinary sized living room. But music is deficient without bass that is supposed to be there, so stereo should have just plain, good bass.

Some of the earlier small bass units relied on harmonic generation to give

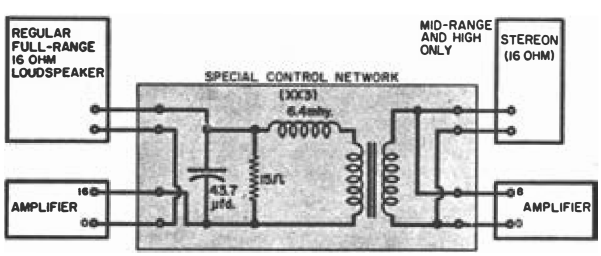


Fig. 7. Connections and network arrangement for the Electro-Voice "Stereon" "add-on" unit is shown in this illustration.

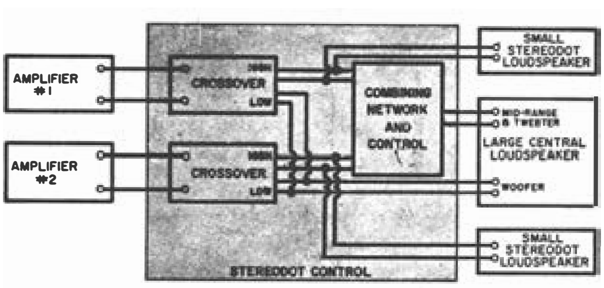


Fig. 8. Block diagram and connections that are utilized in the Stephens' "Stereodot" loudspeaker arrangement. See text.

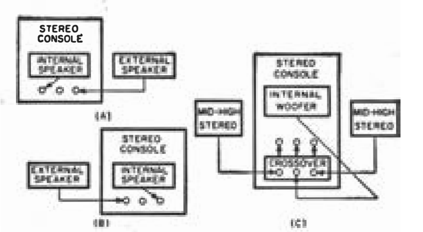


Fig. 9. Arrangements used in Columbia consoles: (A) and (B) use 2 full-range speakers; (C) combines lows in central woofer and mid-range and highs to external units.

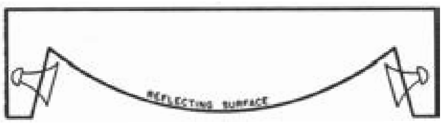


Fig. 10. Basic arrangement used in Ranger-Lansing "Paragon" and "Metregon" units.

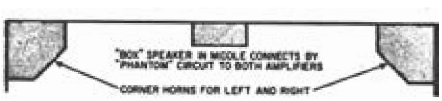
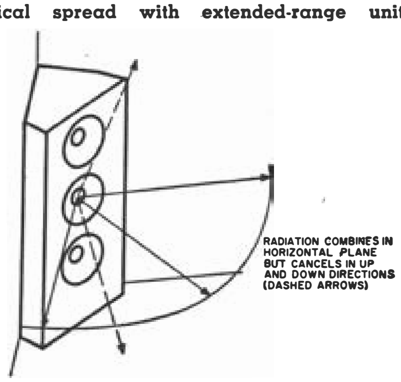


Fig. 11. Klipsch's recommended setup, with 2 corner horns and "phantomed" center unit.

Fig. 12. Method of getting good horizontal distribution and avoiding unnecessary vertical spread with extended-range units.



a "false bass"—like the car radio bass, but more modern approaches have resulted in units that do give real bass from much smaller units than hitherto was believed possible. Some manufacturers claim that trying to get big bass from small speakers is defying "the laws of physics." I think it would be more accurate to say that some designs have found some "loopholes" in these "laws."

But do satisfy yourself that the unit you buy does give genuine, clean bass that is adequate for the room in which you will play it and working from the amplifier you will use.

Two Identical Units

So much for the "system" approaches, some of which come pretty much "ready-made" while others give plenty of scope for experimentation. But another approach that offers wide appeal for two reasons is of the two-identical-unit variety.

This may come either as separate loudspeakers which connect to an equipment cabinet or with the equipment included in one ensemble with a companion speaker to match. In achieving stereo, both offer similar prospects. Also some systems come as a matching set of three which can stand close together thus simulating the one-piece system or be spaced farther out if this is found necessary in a particular listening room.

Directivity

But we now get into another difference over which to choose: directional or diffuse radiating (omnidirectional) type loudspeakers? On this score most speaker manufacturers belong to one school or the other.

The omnidirectional school uses diffuse radiators so that sound exclusively from one unit will be identified with it —wherever you are in the room. Correspondingly different proportions of sound from the two units will be "located" somewhere between, although that "somewhere" may differ with different seating positions relative to the speakers.

A directional loudspeaker is a useful adjunct for improving coverage, especially where provision is made for aiming it, independent of its location, as provided by the *Jensen* "Director" assembly both in separate units and the unitized variety, or by *Goodmans* in its "Stereosphere" (which can be used as part of an installation similar to the "Stereodot").

Directivity of the "stereo" units can be used either to improve the stereo effect in one's favorite listening seat, although the speaker symmetry is not ideal (due to room shape, for example), or to cover a "long shot" position to improve uniformity throughout an area.

One good way of getting a loudspeaker *truly* omnidirectional in the horizontal plane is to "point" it upwards. The *Hegeman*-designed unit put out by *EICO* is an example of this design approach. If the *Goodmans* "Stereosfere" is aimed straight up this does the same. But to give smooth omnidirectional radiation, such units should be close against a wall to avoid undesirable reflection effects from producing effective "double image." This way the reflected radiation merges completely with the direct radiation.

A unit designed to have a highly dispersed radiation from its front only may actually be more uniform in its radiation in some situations—especially if the room arrangement prevents its being close to the wall—due to a radiator or window, for example.

Another factor, besides shape of the area to be covered, is the furnishings although this can cut both ways in different circumstances. In a recreation-type room, with all hard surfaces, a directional pattern that spreads horizontally but restricts vertically is an asset. The *JBL* "Koustical" lens, with its appropriate driver and horn, provides one way of doing this. A less expensive way is to use a vertically aligned row of small direct-radiator units, connected in-phase as shown in Fig. 12. Otherwise, an essentially omnidirectional or diffuse radiation is best for this kind of room.

Rooms with heavy drapes, wall-to-wall carpeting, and heavily upholstered furniture come at the other extreme. Generally speaking, these are more tolerant of loudspeaker types as regards radiation pattern, so concentrate on getting the smoothest response. But sometimes the room shape will tend to produce "dead spots" where the sound gets lost. By "beaming" the sound into these spots with a director system, more complete coverage can be obtained.

That's the story with the well-established and by now familiar *dynamic* speakers, but this year several new names have appeared in the *electrostatic* (I prefer the term "electric"—there's nothing "static" about sound radiation) speaker field. Most of these, thus far, are tweeters or mid-range and tweeter, but at least two claim full-range with electric units only.

Electrostatic Speakers

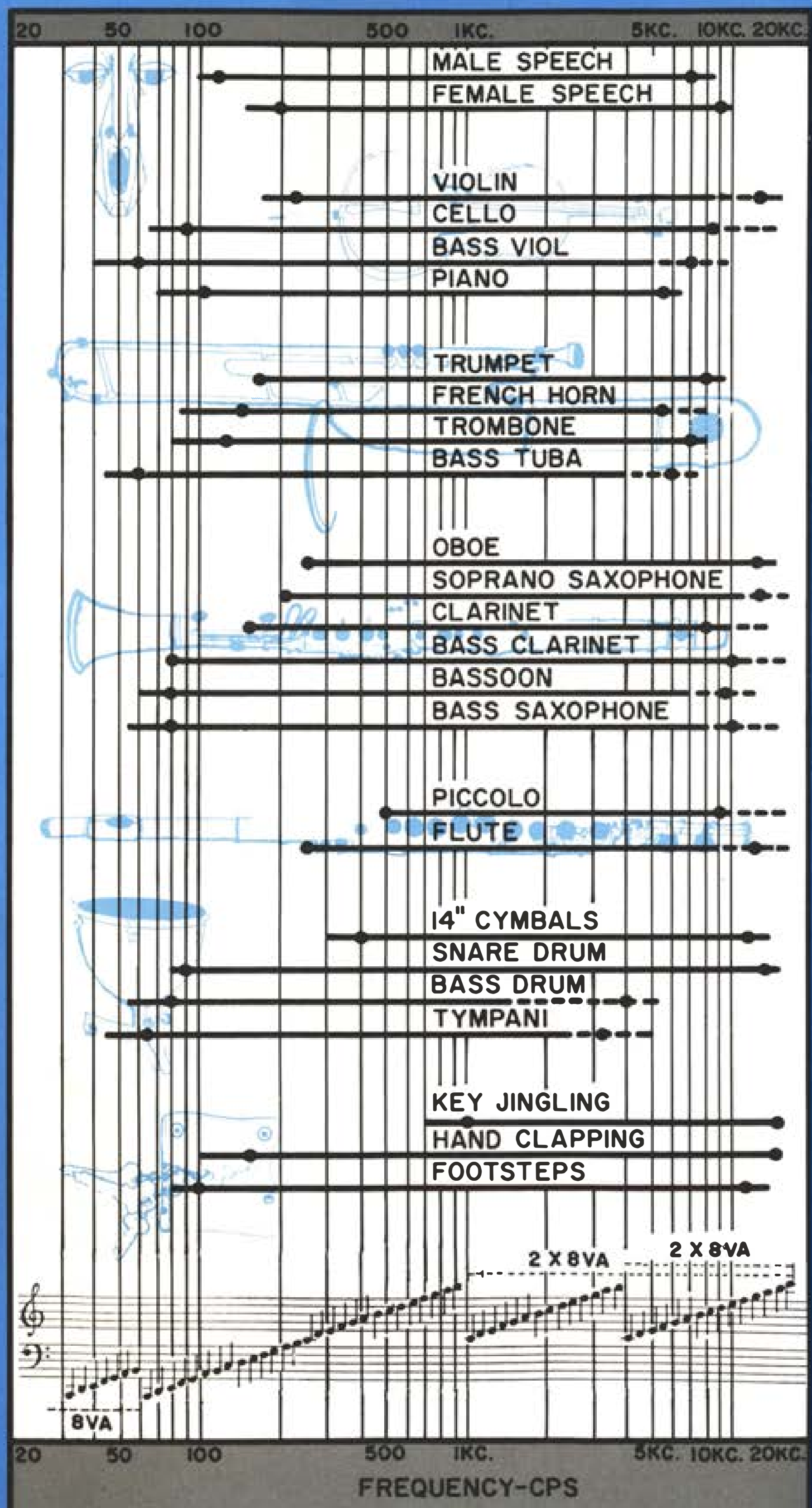
One objection to electric units has been their directional property — a small, flat unit will send out a narrow-angle beam that gets narrower as the frequency gets higher. But this assumes a flat—or near flat—shaping. Actually there seems to be a good reason why this form should be more pliable in making a desired shaping than the more familiar dynamic. Even now, *Electrocoustic* claims a unit that has adjustable directivity by hinging both the radiator and the reflector. The firm says it includes the possibility of omnidirectional radiation. This does provide considerable variation which is a nice feature. In addition, *Wright St. George* is featuring a number of variations including an adaptable "modular" scheme.

A question arises here about the size needed for adequate bass response. This depends on how far the diaphragm can move. Large movement will produce corresponding bass response from a smaller surface area but larger movement requires wider spacing between the fixed and moving elements, a higher impedance amplifier output (which can be provided in a transformer which comes with the speaker), and a very much higher polarizing voltage. This gets into insulation problems which, until recently, have seemed insurmountable. But at last it looks as if a break-through is being made.

When the electrostatic speaker was thought of as essentially a small-movement device, the only way full-range response, including bass, could be visualized was by making at least one whole wall of your room a speaker—which is hardly practical for most of us. That got into a sound radiation concept that was quite the opposite of the conventional dynamic which uses a relatively small cone with large movement. Now the development looks as if it will be quite feasible for electric transducers of the future to optimize on the size question. This may well make them very adaptable units.

AUDIBLE FREQUENCY RANGES

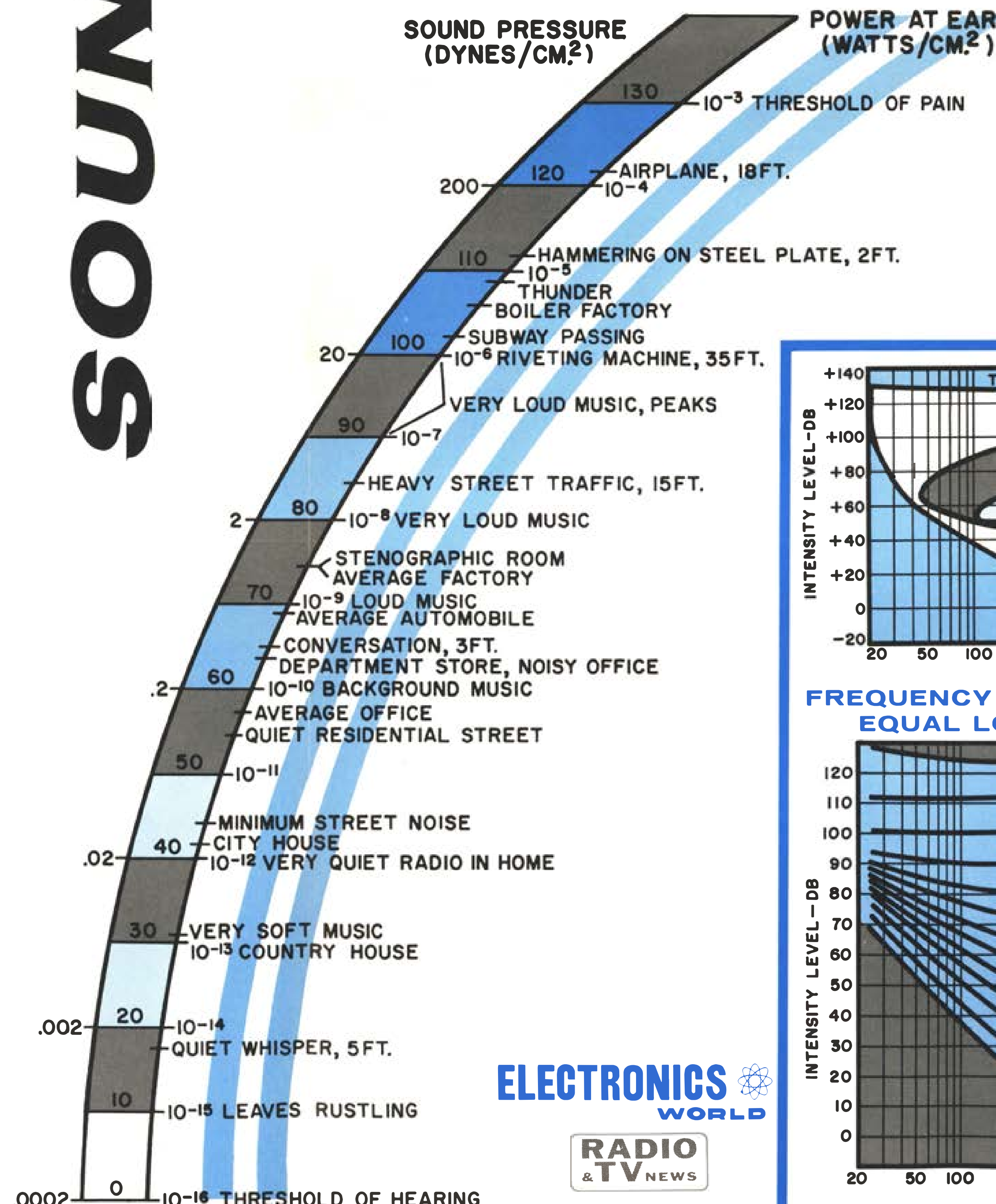
The chart below shows the audible frequency ranges for common musical instruments, speech, and noise. Actual tonal ranges are shown by the solid lines, while accompanying noise range is shown dashed. Points indicated are cut-off frequencies detectable in most tests. Corresponding musical scales are below.



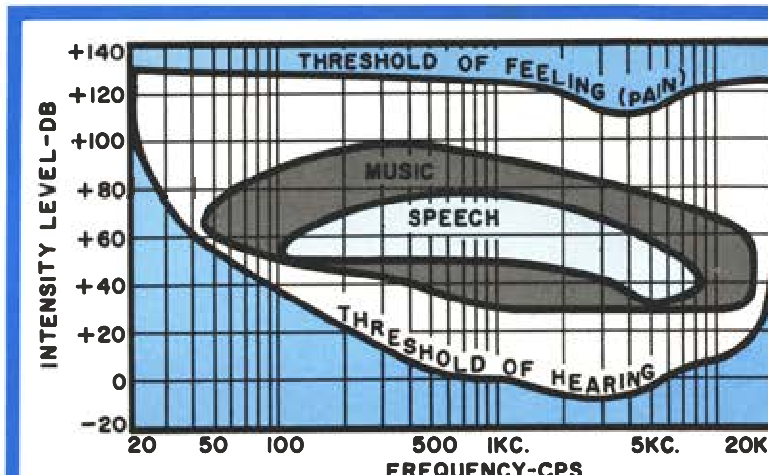
SOUND

SOUND LEVELS

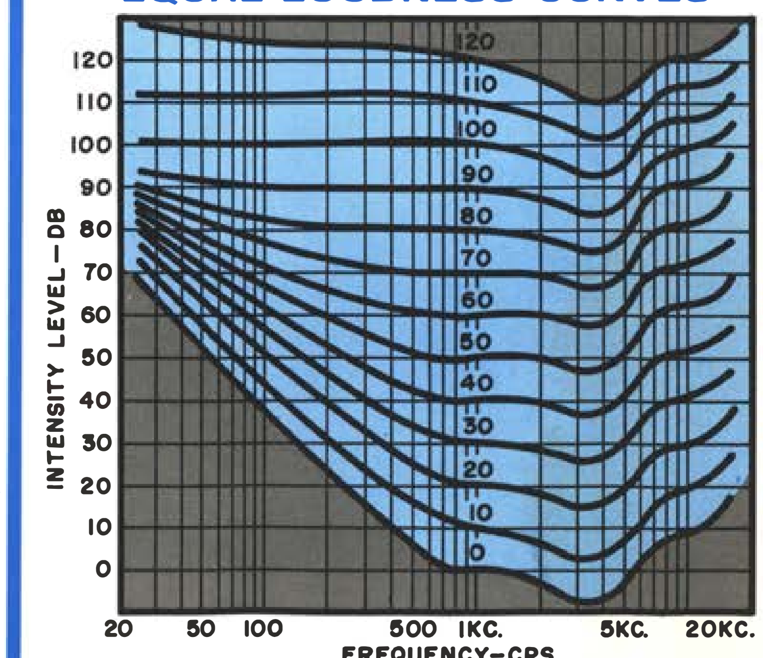
The scale shown below represents the relative levels of various sounds. Note particularly the positions on the scale occupied by music at various levels. In addition to relative level in decibels, corresponding sound pressures (dynes/cm.^{2}) and sound powers (watts/cm.^{2}) are given.



The upper set of curves shows the frequency and intensity ranges of speech and music. Also shown are the boundaries of normal hearing between the two extremes where a sound just becomes audible (threshold of hearing) to where it becomes painful (threshold of feeling). The lower curves show equal loudness contours. Note that as level is reduced, greater intensity is needed at the very low and the very high frequencies to produce equal loudness.



FREQUENCY & INTENSITY RANGES
EQUAL LOUDNESS CURVES

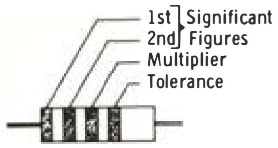


ELECTRONICS WORLD

RADIO & TV NEWS

resistor-capacitor color code

COLOR BAND SYSTEM



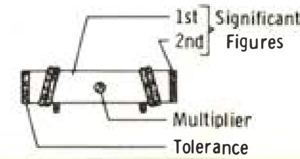
Resistors With Black Body Color Are Composition, Non-Insulated.
Resistors With Colored Bodies Are Composition, Insulated.
Wire-Wound Resistors Have The 1st Digit Color Band Double Width.

RESISTOR CODES (RESISTANCE GIVEN IN OHMS)

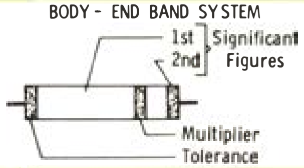
COLOR	DIGIT	MULTIPLIER	TOLERANCE
BLACK	0	1	±20%
BROWN	1	10	±1%
RED	2	100	±2%
ORANGE	3	1000	±3% *
YELLOW	4	10000	GMV*
GREEN	5	100000	±5% (RETMA Alternate)
BLUE	6	1000000	±6%*
VIOLET	7	10000000	±12 1/2%*
GRAY	8	.01 (RETMA Alternate)	±30% *
WHITE	9	.1 (RETMA Alternate)	±10% (RETMA Alternate)
GOLD		.1 (JAN and RETMA Preferred)	±5% (JAN and RETMA Pref.)
SILVER		.01 (JAN and RETMA Preferred)	±10% (JAN and RETMA Pref.)
NO COLOR			±20%

*GMV = guaranteed minimum value, or -0 + 100% tolerance.
±3, 6, 12 1/2, and 30% are ASA 40, 20, 10, and 5 step tolerances.

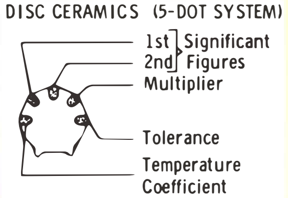
BODY-END-DOT SYSTEM



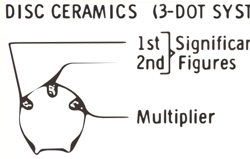
BODY - END BAND SYSTEM



DISC CERAMICS (5-DOT SYSTEM)



DISC CERAMICS (3-DOT SYSTEM)

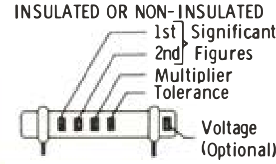


CERAMIC CAPACITOR CODES (CAPACITY GIVEN IN MMF)

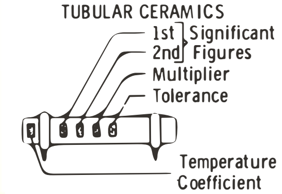
COLOR	DIGIT	MULTI-PLIER	TOLERANCE 10MMF or LESS	TOLERANCE OVER 10MMF	TEMPERATURE COEFFICIENT PPM/°C	EXTENDED RANGE TEMP. COEFF. SIGNI-FICANT FIGURE	EXTENDED RANGE TEMP. COEFF. MULTI-PLIER
BLACK	0	1	±2.0MMF	±20%	0(NPO)	0.0	-1
BROWN	1	10	±0.1MMF	±1%	-33(N033)		-10
RED	2	100		±2%	-75(N075)	1.0	-100
ORANGE	3	1000		±2.5%	-150(N150)	1.5	-1000
YELLOW	4	10000			-220(N220)	2.2	-10000
GREEN	5		±0.5MMF	±5%	-330(N330)	3.3	+1
BLUE	6				-470(N470)	4.7	+10
VIOLET	7				-750(N750)	7.5	+100
GRAY	8	.01	±0.25MMF		+30(P030)		+1000
WHITE	9	.1	±1.0MMF	±10%	General Purpose		+10000
SILVER					Bypass & Coupling		
GOLD					+100(P100) (Jan)		

Ceramic capacitor voltage ratings are standard 500 volts, for some manufacturers, 1000 volts for other manufacturers, unless otherwise specified.

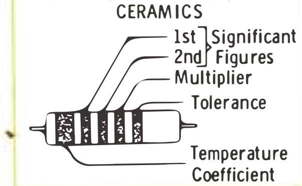
HIGH CAPACITY TUBULAR CERAMIC INSULATED OR NON-INSULATED



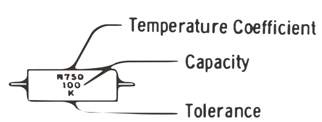
TEMPERATURE COMPENSATING TUBULAR CERAMICS



MOLDED-INSULATED AXIAL LEAD CERAMICS

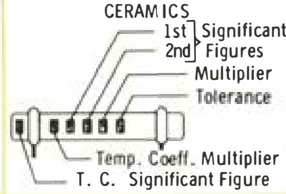


TYPOGRAPHICALLY MARKED CERAMICS



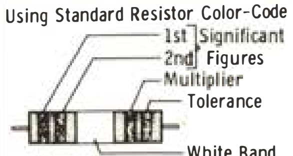
JAN LETTER	TOLERANCE 10MMF or LESS	TOLERANCE OVER 10MMF
C	±0.25MMF	
D	±0.5MMF	
F	±1.0MMF	±1%
G	±2.0MMF	±2%
J		±5%
K		±10%
M		±20%

ENTENDED RANGE T.C. TUBULAR CERAMICS



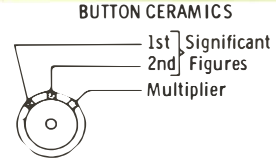
MOLDED CERAMICS

Using Standard Resistor Color-Code



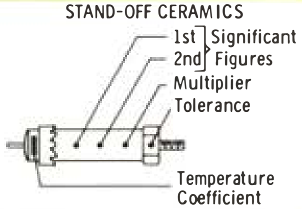
Distinguishes Capacitor From Resistor

BUTTON CERAMICS

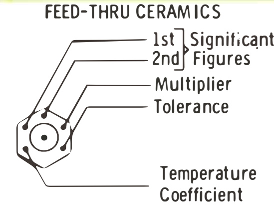


Viewed From Soldered Surface

STAND-OFF CERAMICS



FEED-THRU CERAMICS



MOLDED MICA CAPACITOR CODES

(Capacity Given In MMF)

COLOR	DIGIT	MULTI-PLIER	* TOLERANCE	CLASS OR CHARACTERISTIC
BLACK	0	1	20%	A
BROWN	1	10	1%	B
RED	2	100	2%	C
ORANGE	3	1000	3%	D
YELLOW	4	10000		E
GREEN	5		5%(RETMA)	F(JAN)
BLUE	6			G(JAN)
VIOLET	7			
GRAY	8			I(RETMA)
WHITE	9			J(RETMA)
GOLD		.1	5%(JAN)	
SILVER		.01	10%	

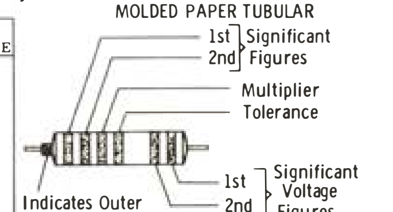
Class or characteristic denotes specifications of design involving Q factors, temperature coefficients, and production test requirements.
All axial lead mica capacitors have a voltage rating of 300, 500, or 1000 volts.
*or ±1.0 MMF whichever is greater.

MOLDED PAPER CAPACITOR CODES

(Capacity Given In MMF)

COLOR	DIGIT	MULTI-PLIER	TOLERANCE
BLACK	0	1	20%
BROWN	1	10	
RED	2	100	
ORANGE	3	1000	
YELLOW	4	10000	
GREEN	5	100000	5%
BLUE	6	1000000	
VIOLET	7		
GRAY	8		
WHITE	9		10%
GOLD			5%
SILVER			10%
NO COLOR			20%

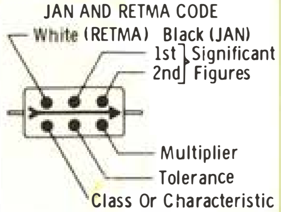
MOLDED PAPER TUBULAR



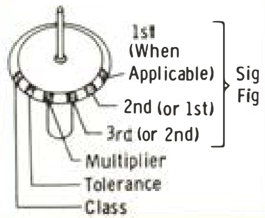
Indicates Outer Foil. May Be On Either End. May Also Be Indicated By Other Methods Such As Typographical Marking Or Black Stripe.

Add Two Zeros To Significant Voltage Figures. One Band Indicates Voltage Ratings Under 1000 Volts.

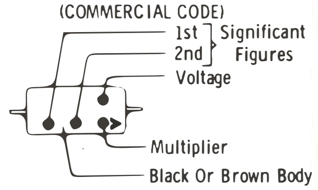
CURRENT STANDARD JAN AND RETMA CODE



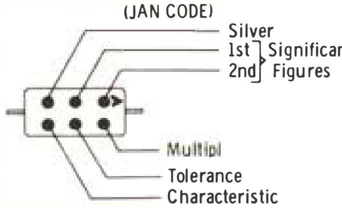
BUTTON SILVER MICA



MOLDED FLAT PAPER CAPACITORS (COMMERCIAL CODE)

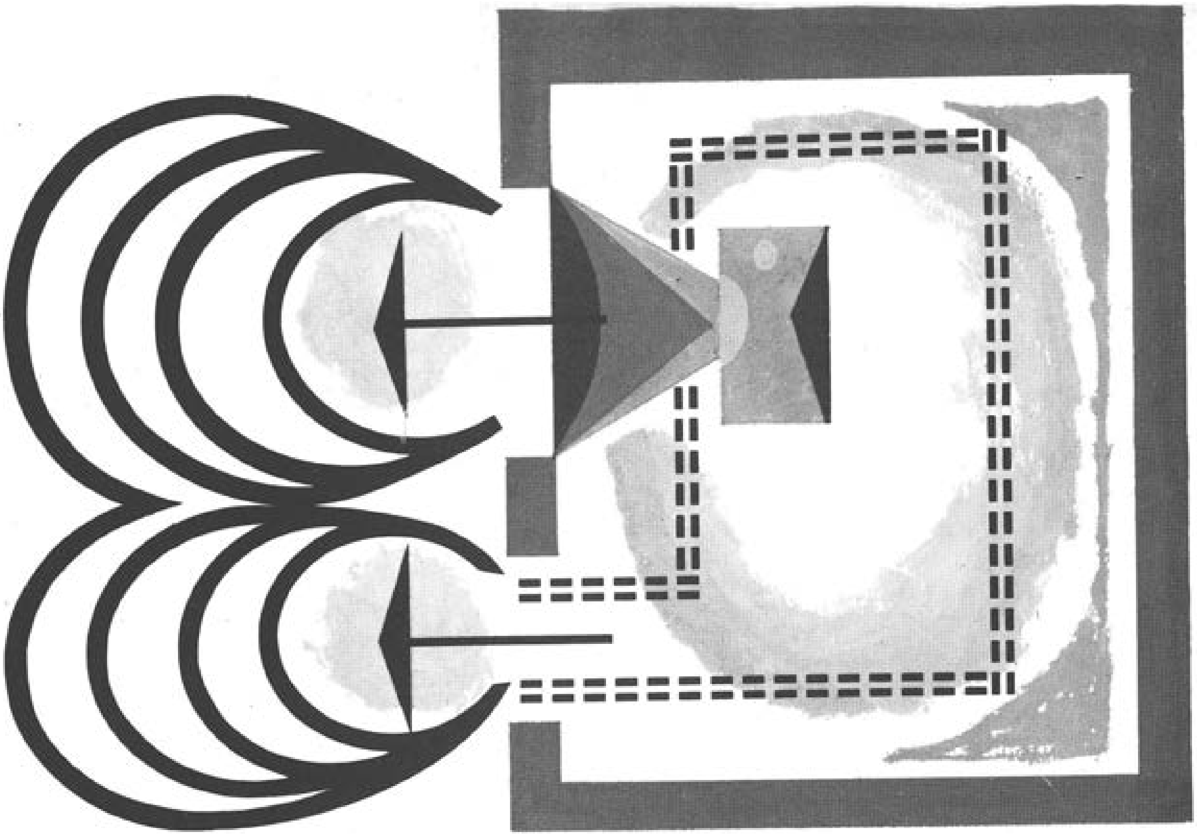


MOLDED FLAT PAPER CAPACITORS (JAN CODE)



DECIBEL TABLE

Voltage Ratio (Equal Impedance)	Power Ratio	← − db + →	Voltage Ratio (Equal Impedance)	Power Ratio
1.000	1.000	0	1.000	1.000
0.989	0.977	0.1	1.012	1.023
0.977	0.955	0.2	1.023	1.047
0.966	0.933	0.3	1.035	1.072
0.955	0.912	0.4	1.047	1.096
0.944	0.891	0.5	1.059	1.122
0.933	0.871	0.6	1.072	1.148
0.923	0.851	0.7	1.084	1.175
0.912	0.832	0.8	1.096	1.202
0.902	0.813	0.9	1.109	1.230
0.891	0.794	1.0	1.122	1.259
0.841	0.708	1.5	1.189	1.413
0.794	0.631	2.0	1.259	1.585
0.750	0.562	2.5	1.334	1.778
0.708	0.501	3.0	1.413	1.995
0.668	0.447	3.5	1.496	2.239
0.631	0.398	4.0	1.585	2.512
0.596	0.355	4.5	1.679	2.818
0.562	0.316	5.0	1.778	3.162
0.531	0.282	5.5	1.884	3.548
0.501	0.251	6.0	1.995	3.981
0.473	0.224	6.5	2.113	4.467
0.447	0.200	7.0	2.239	5.012
0.422	0.178	7.5	2.371	5.623
0.398	0.159	8.0	2.512	6.310
0.376	0.141	8.5	2.661	7.079
0.355	0.126	9.0	2.818	7.943
0.335	0.112	9.5	2.985	8.913
0.316	0.100	10	3.162	10.00
0.282	0.0794	11	3.55	12.6
0.251	0.0631	12	3.98	15.9
0.224	0.0501	13	4.47	20.0
0.200	0.0398	14	5.01	25.1
0.178	0.0316	15	5.62	31.6
0.159	0.0251	16	6.31	39.8
0.141	0.0200	17	7.08	50.1
0.126	0.0159	18	7.94	63.1
0.112	0.0126	19	8.91	79.4
0.100	0.0100	20	10.00	100.0
3.16×10^{-2}	10^{-3}	30	3.16×10	10^{3}
10^{-2}	10^{-4}	40	10^{2}	10^{4}
3.16×10^{-3}	10^{-5}	50	3.16×10^{2}	10^{5}
10^{-3}	10^{-6}	60	10^{3}	10^{6}
3.16×10^{-4}	10^{-7}	70	3.16×10^{3}	10^{7}
10^{-4}	10^{-8}	80	10^{4}	10^{8}
3.16×10^{-5}	10^{-9}	90	3.16×10^{4}	10^{9}
10^{-5}	10^{-10}	100	10^{5}	10^{10}
3.16×10^{-6}	10^{-11}	110	3.16×10^{5}	10^{11}
10^{-6}	10^{-12}	120	10^{6}	10^{12}



Reflex Enclosures

How They Work

By
GEORGE L. AUGSPURGER

Here is a good review of what you should know about an enclosure that's still a favorite for the hi-fi loudspeaker.

MOST loudspeaker enclosures are designed to do three things: (1) achieve full bass response from a given speaker; (2) increase the acoustic efficiency of the speaker; and (3) provide a suitable piece of furniture.

The bass-reflex cabinet is generally claimed to extend the useful range of its cone speaker a half-octave, reduce distortion, and improve transient response. Before investigating these claims, let's briefly examine the fundamental concepts involved in this type of loudspeaker system.

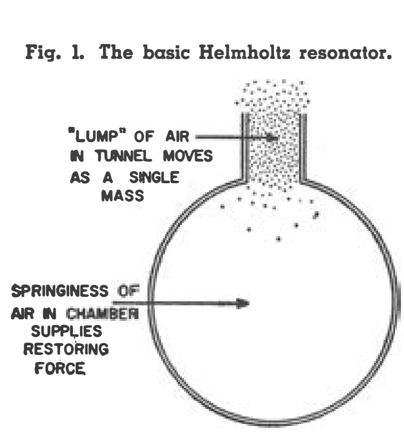
The Helmholtz Resonator

A simple Helmholtz resonator consists of a spherical chamber having a cylindrical spout attached (Fig. 1). The "lump" of air in the spout bounces against the springiness of air in the chamber and the resonator behaves in the same way as a toy whistle or musical jug.

Now, suppose that instead of blowing across the mouth of the jug, we install a piston driven at some audio frequency (Fig. 2). If the frequency of the piston oscillation coincides with the resonant frequency of the original chamber, very small piston movement will produce a considerable amount of sound intensity.

So far, this is all quite simple and easy to visualize. The tricky point is this: at resonance, the piston and the lump of air in the spout are moving in opposite directions—they are out-of-phase. This is not contrary to what we should expect if we remember that the compression and expansion of air in the chamber is an essential factor. If the piston and the air in the spout moved in-phase, there would be no expansion-compression cycle within the chamber . . . the air inside would simply move back and forth.

Fig. 1. The basic Helmholtz resonator.



If this is still confusing, the behavior of such a piston-driven resonator can be verified by experimenting with a simple analogous resonant system. Try hanging a weight from a spring (rubber bands and a table knife work well) and noting the frequency at which the combination oscillates. If you now sustain oscillation by moving your hand up and down, as in Fig. 3, you will find that your hand moves *up* at the same time the weight moves *down*. The driving element (you) and the driven element (the weight) are 180° out-of-phase.

Going back to the acoustic resonator, the elements can be re-arranged as in Fig. 4 without affecting the operation of the system. The piston has been replaced by a cone loudspeaker, the spherical chamber is now cubical, and the speaker and the port are located together on the same side of the chamber. At resonance, the piston and the air in the port move together to alternately expand and compress the air in the chamber. From the standpoint of anyone listening outside the resonator, the port and the speaker are operating *in-phase* to produce sound.

It may seem odd that two elements which were described as being out-of-phase should suddenly turn out to be in-phase after all. The answer is that it all depends on which side of the piston you consider as the source of sound. Since, in practice, *both* sides act as sound sources, we can reverse the radiation from the rear of the cone and use it to re-inforce front radiation.

The Practical Reflex System

The arrangement of Fig. 4 is duplicated in many commercial bass-reflex systems. In practice, the reflex port adds usable radiation in a range extending about an octave on each side of system resonance. One way to get an idea of how such a system operates is to study its impedance curve. Since, in the region below 500 cps, speaker impedance is related to cone movement, an impedance curve gives information concerning the acoustic load which a given enclosure imposes on its speaker. Fig. 6 shows three impedance curves for the same 8-inch speaker. The first is that of the cone speaker in free air, the second is the same speaker mounted in a 2.5-cubic-foot reflex enclosure, and the third curve is of the same combination with the reflex port closed to make a sealed enclosure. Notice that the impedance variation of the reflex system does not exceed a 7:1 ratio, while that of either the free-air speaker or the totally enclosed system exceeds 10:1. Notice also that the minimum impedance of the reflex system occurs at 60 cps—the same frequency as the speaker's free-air resonance. At this frequency the cone is heavily loaded acoustically by the Helmholtz resonance of the system. It is being forced to do work and the increased electrical drain is indicated by a drop in impedance. But since a drop in impedance also means smaller cone excursions, distortion due to magnetic and suspension non-linearity is reduced at the same time.

The sealed enclosure, on the other hand, instead of loading the speaker in the bass region, shifts the impedance peak upwards in frequency. The cone moves farther and farther at progressively lower frequencies and will overload quite easily unless a special long-throw (and relatively inefficient) speaker is used.

This is not to say that the reflex system is inherently superior to a total enclosure. It does explain why speakers designed for reflex loading will not operate as satisfactorily in sealed enclosures or infinite baffles.

Performance

The impedance curves we have just examined bear no direct relationship to frequency response. Unfortunately, the response curves normally published for loudspeakers do not, in themselves, bear much more relation to listening evaluation. Frequency response is merely *one* element in a highly complicated series which determines how closely a speaker system reproduces the real thing.

However, a comparison of the frequency response curves of the same speaker in a reflex enclosure and a sealed enclosure is valuable in further understanding the characteristics of reflex loading. Fig. 5 shows the bass response of the speaker whose impedance is plotted in Fig. 6. These curves were run with the speaker system along the wall of a reasonably large room. Constant voltage was fed to the speaker and a calibrated microphone located three feet from the speaker, on-axis.

Although these graphs clearly show the increase in bass response when a matched reflex enclosure is used, they do not give any information concerning the alleged improvement in transient response. Attempts to measure transient characteristics of loudspeaker systems have been made by various engineers with varying degrees of success. Part of the difficulty is caused by the fact that what the average listener often interprets as "good transient response" may be the exact opposite.

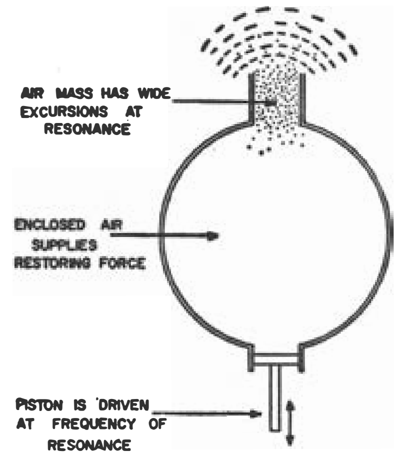


Fig. 2. Resonator driven by a piston.

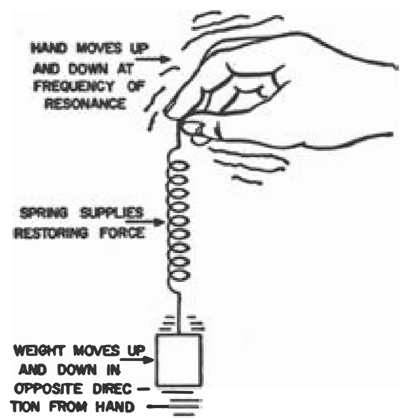


Fig. 3. Resonant system analogous to Helmholtz resonator discussed in text.

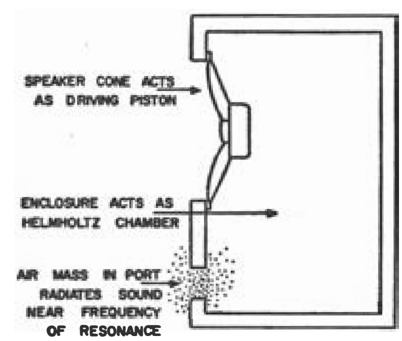
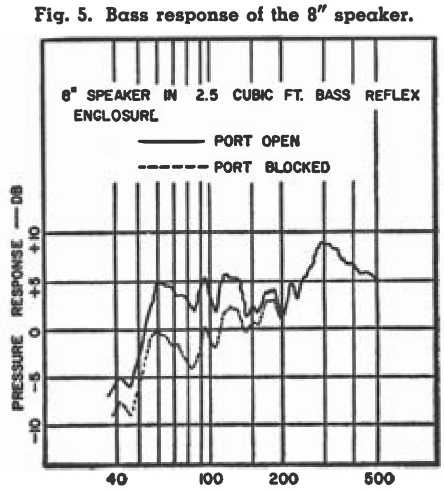


Fig. 4. Standard bass-reflex enclosure.

Fig. 5. Bass response of the 8" speaker.



A full explanation of the factors involved in what is called "transient response" would require several issues of this publication. Generally speaking, if there is any tendency toward "ringing" in the system, this will be audible as a blurring of sharp transient sounds. Moreover, any high-"Q" resonance in the speaker system will be excited by the normal transients in program material. Some speakers tend to make tape hiss sound like turntable rumble.

A totally enclosed loudspeaker has a single resonant peak. This resonance is directly related to impedance and can be controlled to some extent by the electrical damping of the power amplifier. A reflex system, on the other hand, damps the speaker cone acoustically at system resonance.

This feature of high acoustical damping is often cited as a reason why the reflex system should have excellent transient characteristics. Some critics have pointed out, however, that while the reflex enclosure damps the loudspeaker, the loudspeaker does *not* damp the enclosure. And, since the whole thing is basically a Helmholtz resonator, it will take just as long for the signal to decay as if the speaker were replaced with a solid board.

This disturbing state of affairs has prompted numerous methods for "critically" damping reflex systems. Most of these involve the use of acoustic resistance elements in the port. In practice, some efforts along this line result in questionable improvement. The reason is that the Helmholtz resonance is usually tuned to a frequency between 35 and 55 cps and slight ringing in this low range is hardly ever objectionable. The "boomy" quality often attributed to bass-reflex systems is actually not a property of the reflex principle at all, as can be readily demonstrated by blocking the port and noting that the boom is still there.

To get the answer to this puzzle, we must take a look at our impedance curves once again. The upper impedance peak of the characteristic double-humped curve is due to the mass of the cone resonating with a combination of its suspension and the air trapped in the enclosure. This frequency is relatively unaffected if the port is made larger or smaller or sealed up altogether. Fortunately, electrical damping, provided by the power amplifier, helps swamp out this resonance if an efficient speaker is used. It can also be damped by introducing acoustic resistance at the point where air particle velocity is greatest: immediately behind the speaker cone. A partition (as in Fig. 7) made of one-inch acoustical glass fiber is practically a general-purpose cure for bass-reflex systems

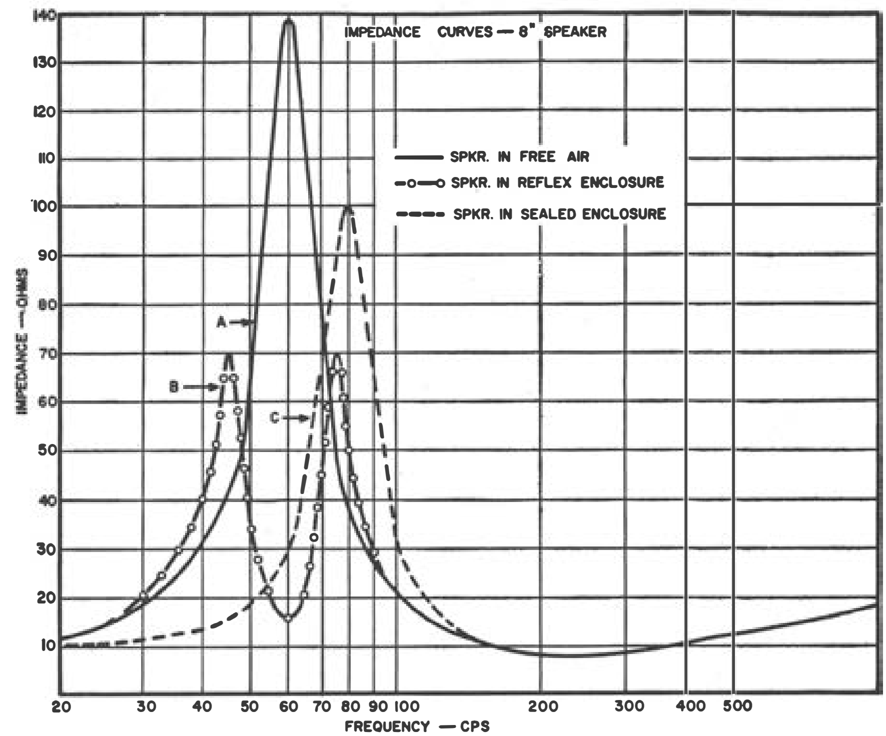


Fig. 6. Impedance curves for 8″ speaker whose bass response is shown in Fig. 5.

in which the upper resonance is pronounced.

Excessive ringing at system resonance, on the other hand, is generally a fault only in very large reflex enclosures. If present, it can be controlled by stretching a screen of light material across the port. Particle velocity is very high in this area at resonance and one thickness of burlap stretched tightly across the opening will usually be sufficient.

To sum up the question of transient reproduction: a bass-reflex system in which the upper impedance peak is controlled and in which system resonance is lower than 55 cps, usually has quite satisfactory transient response. Transient characteristics are further improved if an efficient speaker with high electrical damping is employed.

Designing a Reflex System

Since the bass-reflex cabinet is a Helmholtz resonator, it would seem simple enough to take the basic formula for determining resonant frequency and apply it to the speaker enclosure. Unfortunately, as many experimenters and numerous magazine articles will testify, it doesn't always work.

There are at least four reasons why a practical bass-reflex enclosure deviates considerably from predictions based on the Helmholtz formula:

(1) Interior standing waves introduce peaks and dips which interfere with simple Helmholtz resonance.

(2) Mutual coupling between the port and the speaker cone is a factor not considered in the standard resonance formula.

(3) The walls of any cabinet flex to some degree, introducing still another variable.

(4) It is impossible to guess just which portion of the enclosed air is acting as capacitance and which is behaving as inductance, thus, the shape, position of speaker mounting, and port configuration all influence the final frequency of resonance.

The safest bet is either to use specific recommendations from the manufacturer of a particular speaker or else provide some means of varying the port area and juggle things around until an impedance curve having two equal peaks is achieved.

Rough impedance curves can be run quite easily if a sine-wave generator and v.t.v.m. are available. The circuit of Fig. 8 is usually used. Providing the series resistor is large compared to the speaker's maximum impedance, the voltage measured will be proportional to the impedance of the speaker. However, it is usually not important to be able to calibrate the meter directly in ohms and for the purpose of tuning a reflex enclosure any value of resistance greater than 20 ohms will work.

Merely determining the resonant frequency of the system doesn't even require a meter. A candle held in front of the port while a sine-wave tone is fed to the speaker will indicate quite dramatically the frequency of resonance. But since the exact determination of frequency is not as important as balancing the two impedance peaks on either side to approximately equal values, it is best to use a meter when making any adjustment of a reflex system.

For the home builder who wishes to construct a reflex enclosure without making costly mistakes, the following rules are generally accepted as reliable.

Cabinet Size

Assuming that the enclosure is to be

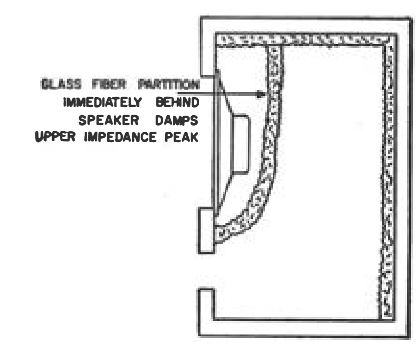
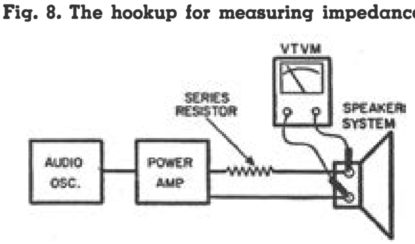


Fig. 7. Reflex system shown with added glass fiber screen to control boominess.

Fig. 8. The hookup for measuring impedance.



tuned to a frequency in the 40-60 cps range, the cabinet is usually made large enough so that the port area will not be less than one-third the cone area of the speaker used. In general, an eight-inch speaker requires an enclosure of 3 to 5 cubic feet, a twelve-inch speaker from 4 to 7 cubic feet, and a fifteen-inch speaker from 5½ to 12 cubic feet.

The shape of the cabinet must not be more than a reasonable departure from a cube. More amateur-designed enclo-

Table 1. Dimensions for building reflex enclosures to house 8″, 12″, 15″ speakers.

8-INCH SPEAKERS (50-60 cps free-air resonance)		8- & 12-INCH SPEAKERS (40-50 cps free-air resonance)		12- & 15-INCH SPEAKERS (30-40 cps free-air resonance)	
CUBIC FEET	PORT AREA	CUBIC FEET	PORT AREA	CUBIC FEET	PORT AREA
2.5	14 sq. in. plus 3″ duct	4	16 sq. in. plus 3″ duct	6	30 sq. in. plus 3″ duct
3	14 sq. in. plus 2″ duct	5	20 sq. in. plus 3″ duct	8	50 sq. in. plus 3″ duct
3.5	16 sq. in.	6	30 sq. in. plus 2″ duct	10	75 sq. in. plus 2″ duct
4	20 sq. in.	7	40 sq. in.	12	85 sq. in.
4.5	25 sq. in.	8	50 sq. in.	14	100 sq. in.

sures probably go wrong on this point than any other—a broom closet with a speaker at one end and a hole at the other simply will *not* behave as a bass-reflex system. The thing to avoid is a cabinet in which the ratio between any two dimensions is more than 3:1. An enclosure 13" x 20" x 50", for example, is obviously a pipe, not a box, and will behave as such.

Reflex Port

The resonant frequency of the enclosure is usually tuned to the free-air resonance of the speaker to be used since this seems to give smoothest bass response. This frequency is determined largely by the volume of the enclosure in relation to the size of the port. It makes little difference if the port consists of one rectangular hole, two slots, or a number of round roles as long as the total area is correct.

Table 1 provides data which can be used to build reflex enclosures with reasonable assurance of satisfactory performance. The table is compiled from the characteristics of actual commercial designs and balances the resonant frequency of the system against the size of the cabinet and the size of the speaker, as well as the speaker's free-air resonance, for best over-all performance.

The exact position of the port or ports is not critical. Theoretically, there is some advantage in having the port close to the speaker, but if it is closer than three or four inches there will be a band of frequencies attenuated because of cancellation between the front and rear of the speaker cone. If close spacing is dictated by the configuration of the cabinet, cancellation can be avoided by inserting a 3 to 4-inch shelf between the speaker and the port on the inside of the enclosure.

A duct or tunnel increases the effective mass of air in the port and thus allows a smaller volume to be used with a given port area. Within limits, this idea works well. The main disadvantage is that even though system resonances can be lowered, the frequency of the upper impedance peak remains very nearly the same. Consequently, if a long tunnel is used in a very small cabinet, there will be an unpleasant boom in the upper bass region and a hole in the response curve below this point.

Construction

In even the most rigidly constructed cabinet, there is always some flexion of panels. The ideal method of building a reflex enclosure seems to be to use brick or concrete. For most applications however, wood is still the most practical material, and a carefully built wooden cabinet can closely approach theoretically optimum performance.

The enclosure should be made of plywood ¾- to one-inch thick and any panels larger than two feet square should be braced internally. If there is noticeable panel vibration once the system is in use, more struts should be added until all surfaces are rigid and free from resonance.

Interior Padding

Considerable misunderstanding exists concerning the function of acoustic padding in reflex cabinets. Contrary to published information, varying the amount of interior padding over wide limits will not noticeably change the resonant frequency of the system nor will it affect bass transient response.

The real purpose of lining the interior walls with absorbent material is to smooth out mid-range response. The more padding used, the smoother the measured frequency response, but the "deader" the mid-range will sound. Obviously, the correct amount of treatment depends on the mid-range characteristics of the speaker, the acoustics of the room in which the system is used, and the degree of liveness which the listener finds natural. A general rule of thumb is to pad four interior surfaces and arrange to have each blank wall face a padded wall.

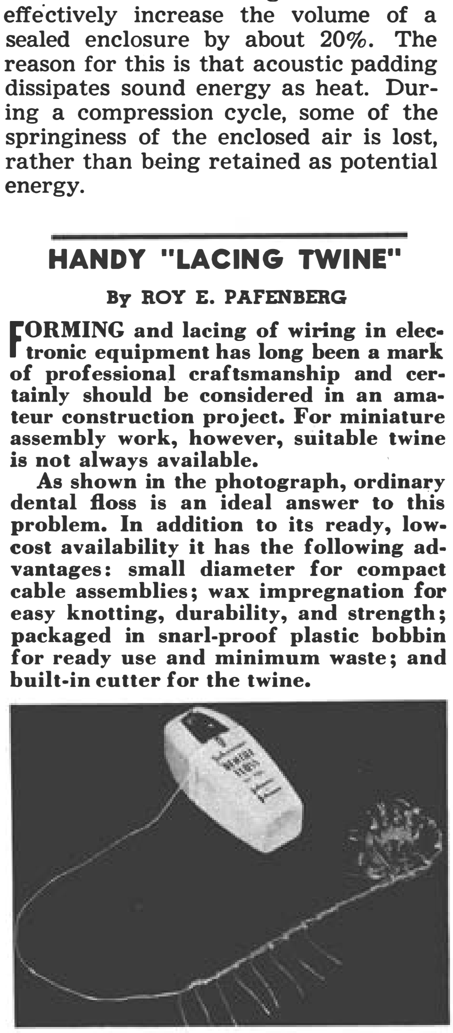
In computing the volume of a reflex enclosure, the space occupied by padding should not be subtracted from the internal dimensions. As a matter of fact, if a great deal of absorptive material is used, the effective volume of the cabinet in increased rather than decreased. Lining all surfaces with four or five inches of glass fiber will effectively increase the volume of a sealed enclosure by about 20%. The reason for this is that acoustic padding dissipates sound energy as heat. During a compression cycle, some of the springiness of the enclosed air is lost, rather than being retained as potential energy.

HANDY "LACING TWINE"

By ROY E. PAFENBERG

FORMING and lacing of wiring in electronic equipment has long been a mark of professional craftsmanship and certainly should be considered in an amateur construction project. For miniature assembly work, however, suitable twine is not always available.

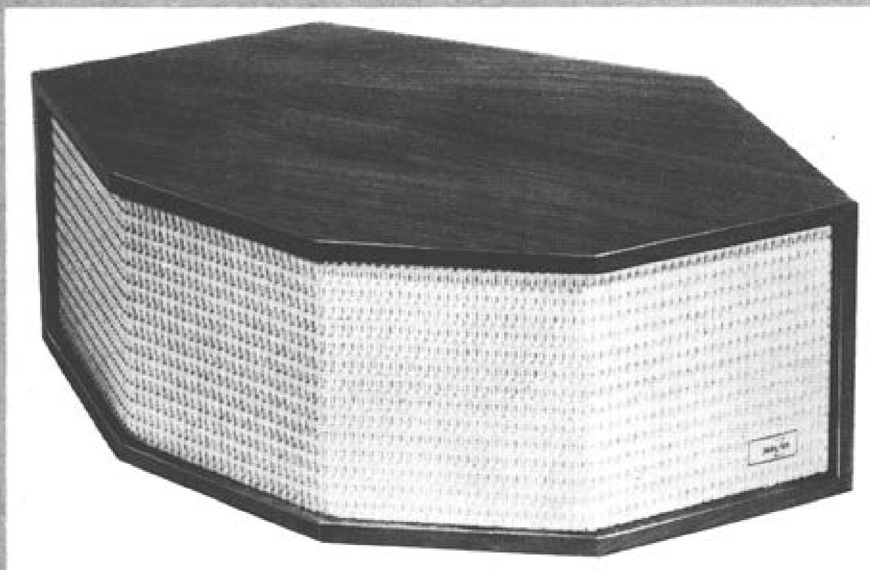
As shown in the photograph, ordinary dental floss is an ideal answer to this problem. In addition to its ready, low-cost availability it has the following advantages: small diameter for compact cable assemblies; wax impregnation for easy knotting, durability, and strength; packaged in snarl-proof plastic bobbin for ready use and minimum waste; and built-in cutter for the twine.



Testing Electrostatic Loudspeakers

By ARTHUR A. JANSZEN

Listening tests versus laboratory measurements—what's the right way to assign a figure of merit to a hi-fi speaker?



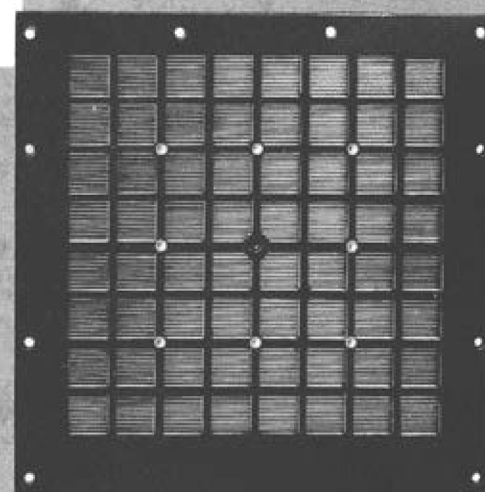


Fig. 1. JansZen Model 130 (above) uses 4 radiators of type shown (right).

IT APPARENTLY is still fashionable to refer to loudspeakers as the "weakest links" in the chain of components in high-fidelity systems. The obvious inference to draw from such statements is that in every high-fidelity system, the loudspeaker is the most deficient of all components in the performance of its assigned role. This is not true. There are *some* loudspeakers that introduce less over-all distortion than *some* pickup cartridges, tape playback systems, preamplifiers, power amplifiers, and tuners. Despite the effort that has gone into investigations of the design parameters that make one loudspeaker "good" and another "bad," the problem of loudspeaker evaluation is still with us; for there is no measurement or set of measurements that can be used to predict accurately whether a "typical" listener will prefer speaker *A* over speaker *B*. It may be possible in the future, after extensive bio-acoustical experiments have been completed and the data evaluated, to make a prediction of listener preference, based on objective measurements, that will be valid for *most* listeners in *most* situations. At the present time, however, such a statistical prop is not available and there is only one way to find out whether speaker *A* or speaker *B* is to be preferred in a given situation, and that way is to try *A versus B* under the conditions in which listening is to be done.

Listening Tests

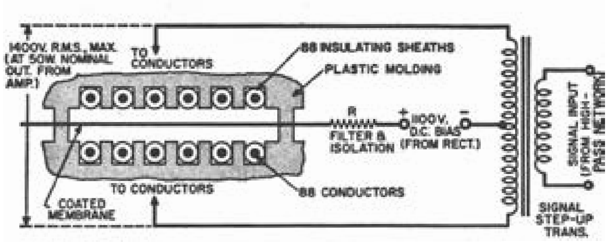
The assignment of a "figure of merit" to a loudspeaker can be valid only under highly specific conditions. This is because one's listening response depends not only on the speaker's actual performance, but also on the acoustical environment in which the listening is done, on the performance of associated equipment, and on subjective factors that defy definition. Judgments of loudspeaker performance based on listening tests can be valid only for one panel of listeners, in the particular listening room, for the particular positions of the listeners and loudspeakers within the room, and for the particular program material used, in conjunction with a particular set of associated equipment. For example, a preference for one loudspeaker over another can sometimes be reversed by simply using a different amplifier.

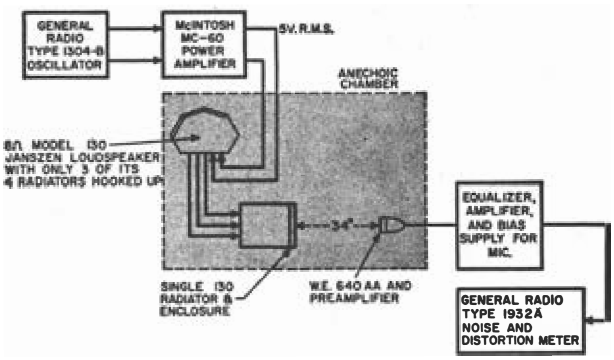
Table 1. Total harmonic distortion in acoustic power output of a single unselected radiator at various frequencies with 5 volts applied to the 8-ohm input.

FREQUENCY (in cps)	% TOTAL HARMONIC DISTORTION
750	0.47
1000	0.4
1500	0.35
2000	0.23
4000	0.16
5000	0.38
6000	0.47
7500	0.5
10,000	0.45

Objective Measurements

Although objective measurements do not permit the assignment of definitive figures of merit to loudspeakers, there *are* several performance factors, which *are* susceptible of objective measurement, that are important in determining critical listener preference, even though they may not represent *all* of the pertinent factors. These are: (1) the range of frequency





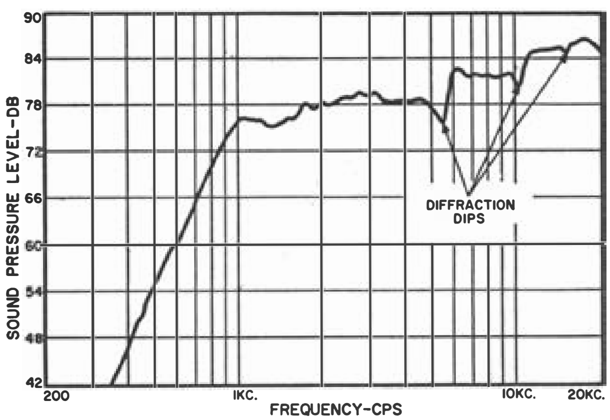


Fig. 2. (Top left) Electrical connections and construction of electrostatic radiator described in text. Circuit for the 1100-volt bias supply (derived from the a.c. line) and the high-pass LRC filter network required is not shown in drawing.

Fig. 3. (Top) Test setup for distortion measurements. The radiator being checked was mounted in a separate enclosure one-quarter the volume of main enclosure. All 4 radiators were connected for proper amplifier loading but radiation from the 3 radiators not under test was prevented by disconnecting bias.

Fig. 4. (Left) Axial pressure response of a single radiator with 2 volts input at 8 ohms. Note slight diffraction dips.

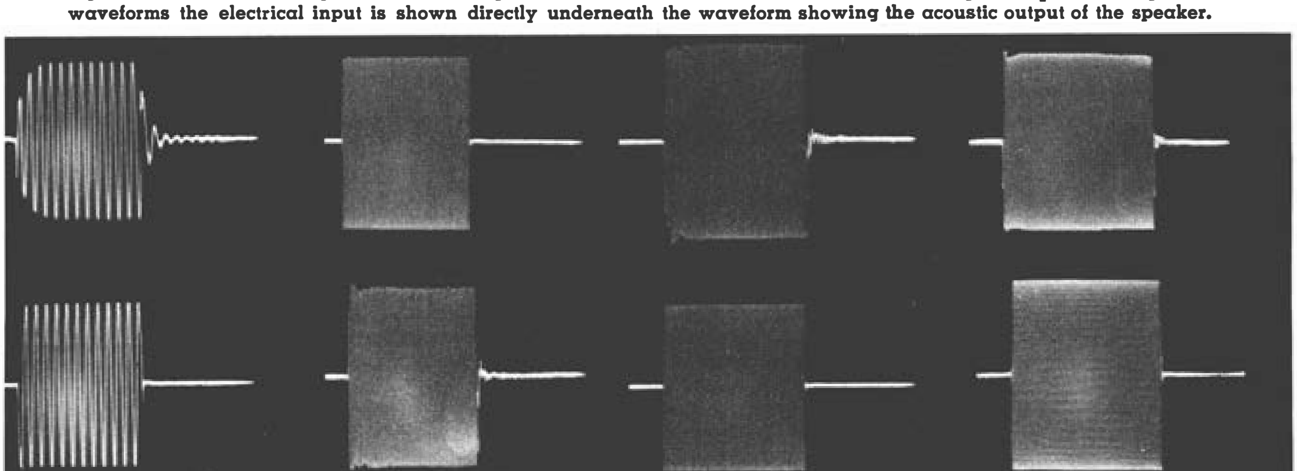
response; (2) the "trend" or shape of the curve of the frequency response curve, *i.e.,* whether some bands of frequencies are emphasized or de-emphasized with respect to other bands; (3) the "smoothness" of the frequency response curve, *i.e.,* the presence or absence of sharp dips and peaks in the curve; (4) the linearity of response, *i.e.,* whether a linear relationship exists between the instantaneous values of input voltage and output pressure over the entire a.c. signal cycle at each frequency within the bandpass, which can be determined by a measurement of total harmonic distortion; (5) the transient response of the system, which can be inferred from the frequency response but which can be more directly investigated by applying "tone bursts" of various carrier frequencies within the passband to the input terminals and photographing the output of a microphone as oscillograms; (6) the distribution of acoustic pressure in both vertical and horizontal planes as a function of the angular position with respect to the axis of the loudspeaker; and (7) the impedance frequency characteristic, which affects the ability of the associated power amplifier to supply the required output voltage frequency characteristic at the required voltage levels.

These then represent the objective measurements that can be made.

It must be stressed that the listener's ability to detect differences in performance between one loudspeaker and another, will depend greatly on the performance of the associated equipment. It is incumbent upon the manufacturer of loudspeakers that are capable of excellent performance to be specific in making recommendations concerning the other components in the system. For example, if another component (or components) is generating a lot of harmonic distortion at 4000 cps, it is likely that a loudspeaker whose response is restricted to frequencies below 8000 cps would be chosen over one whose response extends to 20 kc.

Since a loudspeaker should convert complex electrical waveforms into acoustical counterparts without distortion, it would seem desirable to make this conversion *directly,* without membranes, cones, domes, or horns. Such a system would be of ultimate simplicity. If the acoustic output varied directly with the input voltage, and if the area of the air front at the radiating boundary were large enough to prevent non-linear response of the air itself, then one would have a loudspeaker free of waveform distortion. If the area were made appropriate to the range of frequencies to be radiated, then the power output could be made independent of frequency (for a constant input voltage), and the electro-acoustical tran-

Fig. 5. Tone bursts at a repetition rate of 30 per second for 1 kc., 3.5 kc., 7 kc., and 20 kc. respectively. In each pair of waveforms the electrical input is shown directly underneath the waveform showing the acoustic output of the speaker.



1 KC. **3.5 KC.** **7 KC.** **20 KC.**

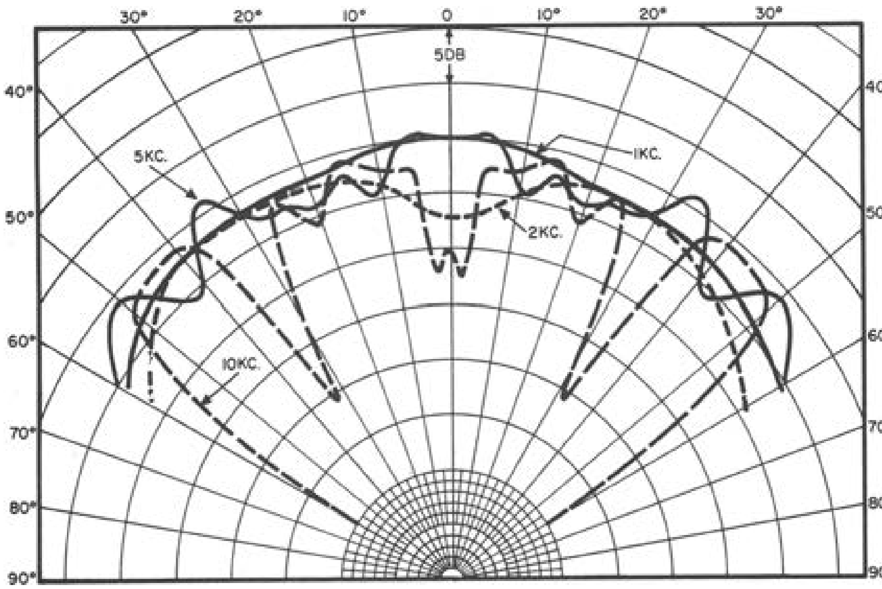


Fig. 6. Polar charts of acoustic pressure at various frequencies with all 4 radiators operating. Measurements were taken in anechoic chamber at distance of 10 ft.

sient response would be without flaw.

Such an ideal loudspeaker is not yet practical and all speakers on the market at present have one or more mechanical and/or acoustical coupling elements which tend to introduce resonances and which, in turn, cause a degradation of transient response. In the radiating elements used in the speaker shown in Fig. 1, an attempt has been made to achieve the greatest degree of simplicity and the nearest approach to the ideal electro-acoustical system. These elements are push-pull, constant-"*q*" electrostatic radiators, in which the vibrating elements are made of plastic membranes so light and thin that over most of their frequency range they operate almost as if the membrane were absent. Through the application of a very high, constant d.c. charge ("*q*") between the membrane and the two stationary electrodes, a high degree of linearity can be achieved with a high per-unit-area acoustic power output.

Test Results

Fig. 2 shows the electrical connections to a segment of a radiator. Signal voltage is applied to the outer electrodes and bias voltage is applied between the conducting coating of the membrane and the outer electrodes by means of a resistor *R*. An analysis of this constant-"*q*" system, in which the electric charge deposited by the bias supply is kept constant during variations in signal voltage by the presence of the high resistance, leads to the conclusion that if perfect symmetry is preserved, there is no harmonic distortion. If *R* is made large enough, electrode asymmetry, within limits that can be maintained in production, causes only very small amounts of distortion. The degree of excellence with respect to distortion that can be achieved in production is shown in Table 1.

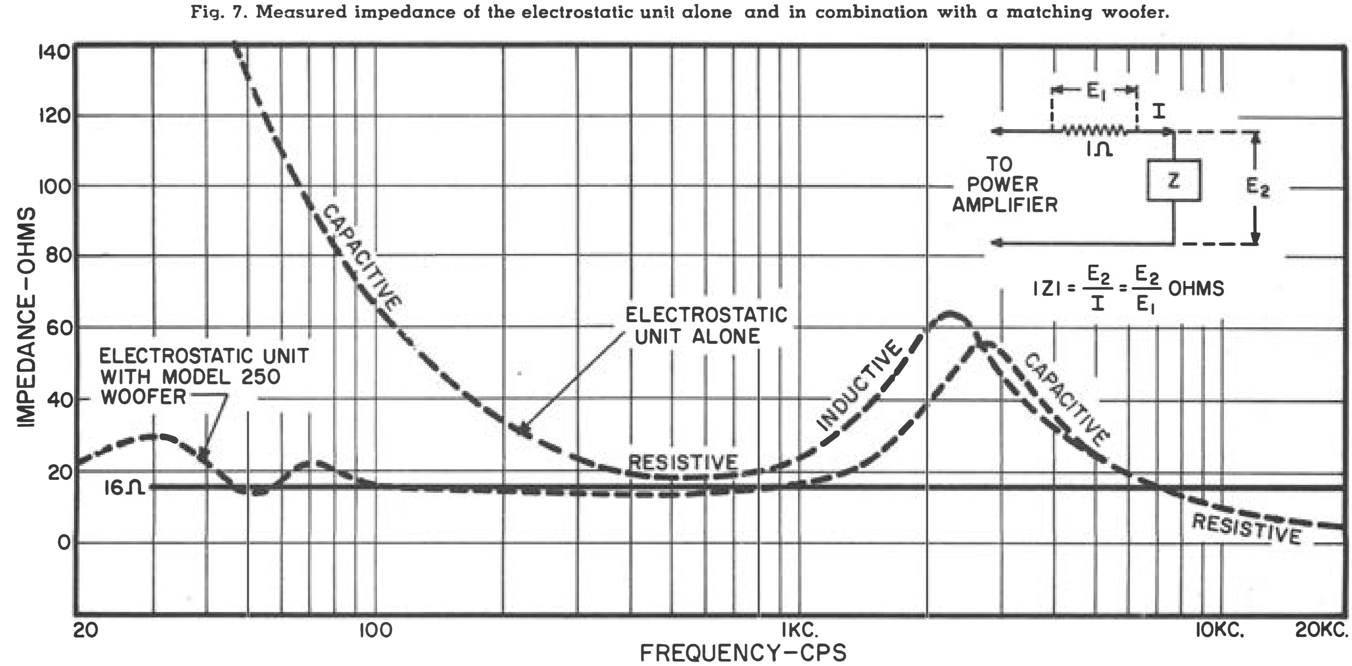
Fig. 3 shows a diagram of the test setup used in the measurements. Data gathered in an anechoic chamber on a multiple array of radiators defies interpretation, so measurements were made on a single radiator. The response of the microphone used was equalized by means of a multiple *LRC* network so that the over-all system response was flat to within 1 db up to 20 kc.

A pressure frequency response curve, taken on axis, is shown in Fig. 4. This was obtained with the test setup of Fig. 3, except that a power level recorder, mechanically linked with the oscillator, was substituted for the distortion meter. The rising pressure response tends to keep the acoustic power output more nearly constant than it would be if the curve were flat. The average reverberant living room translates this rising pressure characteristic, which is accompanied by increased directivity, into a more nearly constant pressure response.

Although the transient response of a loudspeaker can be inferred from its pressure frequency response, tone bursts provide a more direct test. Fig. 5 shows oscillograms of bursts of various carrier frequencies within the passband of the radiator. As one would expect, the 1000 cps bursts, at the lower end of the passband, were not as good replicas of the electrical signal as the remainder of the bursts. The absence of "hangover" is indicative of the effectiveness of a "nearly absent" vibrating element.

Fig. 6 shows the acoustic pressure at various angles off-axis of a Model 130 with all four radiators in operation, at several frequencies. At frequencies below about 8 kc., the response in the horizontal plane is quite uniform over a total angle of about 120°. At 10 kc. and above, there are sharp dips in the

Fig. 7. Measured impedance of the electrostatic unit alone and in combination with a matching woofer.



polar pressure between the beams of the individual radiators. The reverberation of the listening room must be relied upon to smooth these out.

The impedance-frequency curve is important in determining the "power" transfer of an amplifier for a given distortion. Maximum undistorted power transfer can be attained generally when the amplifier load is resistive and equal to the nominal output impedance of the amplifier. The loudspeaker carries nominal impedance ratings of 8 ohms or 16 ohms. The actual impedance, as in most loudspeakers, deviates considerably from the nominal rating. Fig. 7 shows the measured input impedance of a randomly selected unit. The impedance is near the rated value in two frequency regions, around 600 cps and around 7000 cps. At the lower frequency, the impedance is primarily resistive, while at the higher frequency it is primarily capacitive.

To obtain the desired pressure response from the unit, as indicated in Fig. 4, it is essential that the signal voltage applied to the radiators be constant within the passband. Therefore, only power amplifiers with a high damping factor (low source impedance) are recommended. Such amplifiers deliver a constant voltage to the input terminals of the speaker over a wide frequency range. Speaker impedance variations then are important only as they affect the ability of the amplifier to deliver undistorted and constant voltage. The undistorted power that a good amplifier can deliver into a 2-ohm resistive load at 20 kc. from its 8-ohm terminals is usually not greater than about 10 per-cent of its rated power output. But since the power-per-cycle in music is maximum in the region between about 100 and 500 cps and falls off rapidly above 1000 cps, the impedance mismatch that occurs at the extreme high end of the hearing range does not normally degrade the performance.

Since only the upper 50 per-cent of the musical scale is reproduced by the electrostatic unit, it must be used with a low-frequency loudspeaker, or woofer. Several are readily available. The power transfer capabilities of the amplifier are more important for the combination of woofer and tweeter than for the tweeter alone, since it is the combination that acts as load on the amplifier. Fig. 7 also shows the combined impedance of a typical 16-ohm dynamic woofer with the tweeter. The upper end of the impedance curve is almost identical with the curve for the electrostatic unit alone, while in the region of maximum music power density (100 to 500 cps), the impedance is within the limits of 13.2 and 16 ohms.

The electrostatic radiators described in this article are covered by patents issued and pending to the author.

Simplexing For Low-Level Crossover

A single push-pull stage is able to handle both bass and treble channels with little interaction.

By **C. NICHOLAS PRYOR**

RECENTLY the idea of placing the crossover network of a multi-channel loudspeaker system in front of the power amplifiers has gained many adherents. The advantages of such a scheme are generally conceded and will not be discussed here. The disadvantage of cost, however, has deterred many people from adopting low-level crossovers. With the method to be described here, the cost of a low-level system is on a par with, if not less costly than, an equivalent high-level crossover.

The method involves a "trick" long used by telephone and broadcast engineers—namely, simplexing. Basically, the idea consists of feeding an unbalanced signal down a balanced line along with the balanced signal, thus providing two communication paths along the same line. It has been applied to audio amplifiers,[1] at least once, to permit use of the same amplifier for two programs and works as follows: Consider the circuit shown in Fig. 1. It represents a push-pull stage in an audio amplifier. With no voltage drive on the grids a static current flows through ammeter M_2 and no net voltage is shown across the voltmeter, M_1. If the grids are driven equally in opposite directions (push-pull) and assuming linear operation of the tubes, a voltage appears across M_1 but the current in M_2 remains the same. On the other hand, if the two grids are driven equally in the same direction, the current of M_2 will change in proportion to the grid voltage while no net voltage appears across M_1.

By superposition, then, we can drive the grids both in push-pull and in parallel with different signals and have these signals appear individually on the two meters. By replacing the say that this makes the system impracmeters with transformers, we now have a power stage for the amplifier that will handle two *different* signals *independently*. If the signals are unrelated, very precise balance and low distortion must be maintained (both under .1% for 60 db isolation of the two signals). One would be likely to tical but there is one important instance where it is very practical indeed. This is the case when the two signals are the high- and low-frequency ends of the same program material! Here it is only necessary to have about 30 db isolation between signals and this is readily obtainable with the simplex arrangement (this allows up to 3% unbalance and distortion).

One of the worthwhile features of simplex is that it may be installed in practically any class A amplifier at nominal cost. Fig. 2 is a partial schematic of a typical Williamson-type amplifier with the simplex feature added. The dotted block indicates the modifications to be made on the existing circuit and below the dashed line are the additional circuits required. If desired, these may be built on a separate chassis and mounted near the main amplifier.

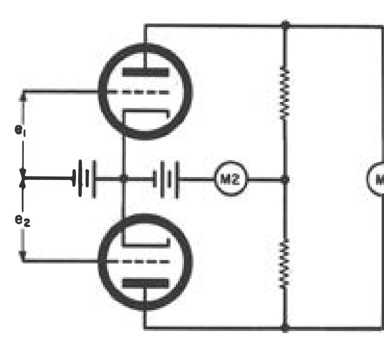


Fig. 1. Basic circuit arrangement used.

The author's model uses a pair of 6BX7's for finals and puts out 8 watts from the balanced section and 6 watts from the simplex. Mr. Augspurger[2] states that equal powers are needed on both channels for a crossover of 400 cps, but this statement must be qualified. It assumes that the tweeter and woofer have equal power sensitivity, which is not generally the case. Most horn-type mid-range or tweeter units are from 3 to 10 db more sensitive than bass drivers in their usual enclosures. This means that from 1/10 to 1/2 as much power will be needed on the tweeter as on the woofer and the simplex handles this adequately.

When selecting a transformer for simplex it is not necessary to buy a lot of steel. This channel is for the high end so the primary inductance (and thus the weight and price) of the transformer need not be too large. It is necessary, though, to buy a unit that will handle the unbalanced current that will flow through it. The best thing to do is buy a single-ended unit that is rated at about two-thirds the current rating of the power transformer used in the amplifier. It might also be necessary to increase the filtering of the power supply. Such filtering is usually designed for a balanced power stage, so the hum voltage on the "B" supply might be too great for single-ended operation without extra filtering.

As for the crossover itself, any standard design will do. The author's circuit is shown in Fig. 3A. It is a constant-resistance *LC* circuit using plug-in components for variable crossover frequency. Since audio chokes are expensive, the simple *RC* circuit of Fig. 3B may be used. Equations for the design of both circuits are given on the diagrams. The low-frequency output of the crossover is connected to the original amplifier input and the high-frequency crossover output is connected to the point marked "HF in" in Fig. 2.

That is all there is to it! The cost of simplexing a 10-watt amplifier, including the cost of the *RC* crossover, is about $10.00 as against approximately $20.00 for high-level crossovers. It's inexpensive but, most important of all, it sounds great!

Fig. 2. Circuit showing connection of simplex to a typical push-pull amplifier.

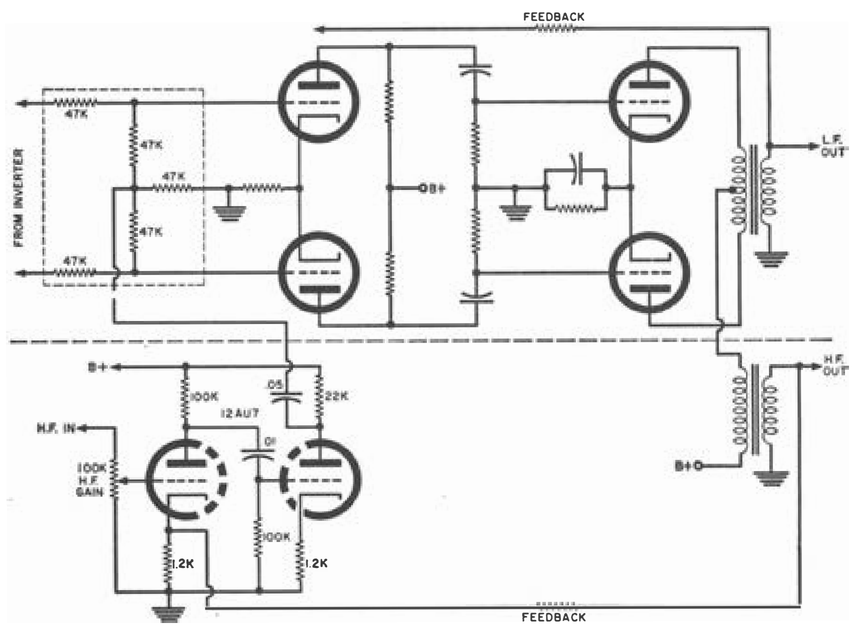
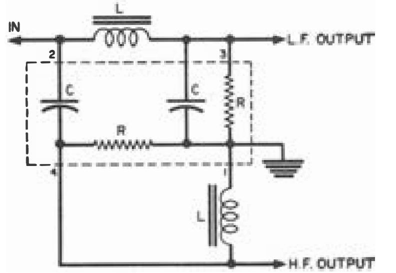


Fig. 3. (A) 12 db per octave LC network. (B) 6 db per octave RC crossover network.



L=22hy. THORDARSON 20C74

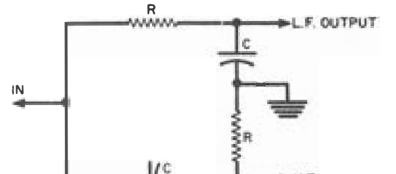
$R' = \sqrt{2}\pi f L$ OHMS

$C = \frac{1}{2\sqrt{2}\pi f R'}$ FARADS

R'=R IN PARALLEL WITH AMPLIFIER INPUT RESISTANCE

f=CROSSOVER FREQUENCY IN CPS

(A)

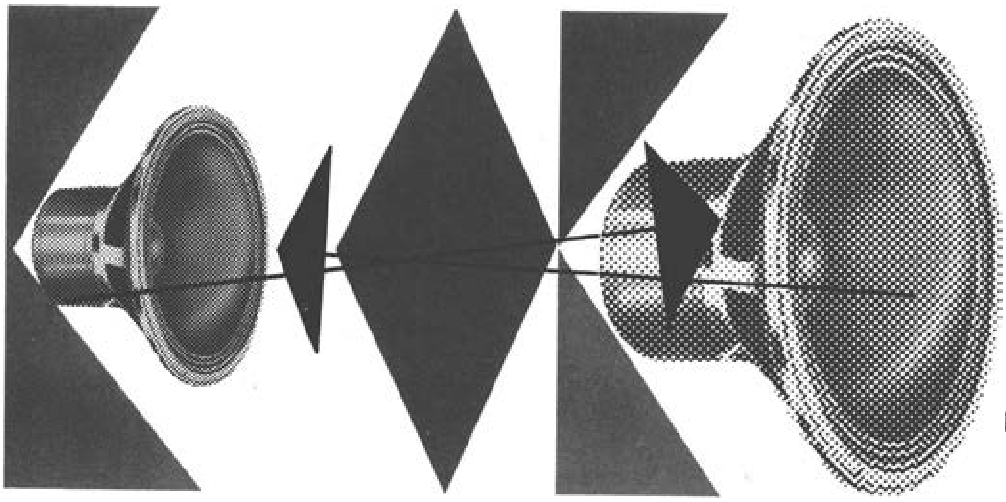


$C = \frac{1}{2\pi f R'}$ FARADS

R'=R IN PARALLEL WITH AMPLIFIER INPUT RESISTANCE

(B)

HI-FI Crossover Networks



By ABRAHAM B. COHEN &
PAUL D. COHEN

Part 1. Here are the facts you need to know about these important circuits before building or buying.

AUDIO dividing networks are essential components that ensure the proper functioning of multi-speaker systems. A well-designed audio dividing network, or crossover network as it is usually called, performs two functions. First, it is a traffic policeman that directs the various parts of the audio spectrum to the specialized speakers which are best able to handle specific bands such as the lows and highs. The secondary function of the crossover network is to protect the delicate tweeter mechanisms from low-frequency overload. The end *result* of these combined functions is better utilization of audio power available from the amplifier, cleaner reproduced sound from the loudspeaker, and more direct control over what comes out of the loudspeakers.

There are simple networks and there are complex networks, but they are all easily understandable when approached in a basic fashion. Before going into details of the design and construction of home-built precision networks, which will be covered in Part 2, a simple and quick recapitulation of the principles behind the network function will help the builder decide what network he should construct for his system.

Speakers Without Network

Networks, of course, are used with multi-speaker systems. The simplest multi-speaker system consists of a woofer for the reproduction of low frequencies and a tweeter for the reproduction of high frequencies. Although it would be unrealistic to connect two components such as a woofer and a tweeter directly to an amplifier without benefit of a network, we will do just that to illustrate what happens to

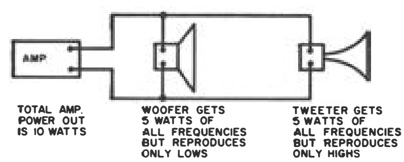
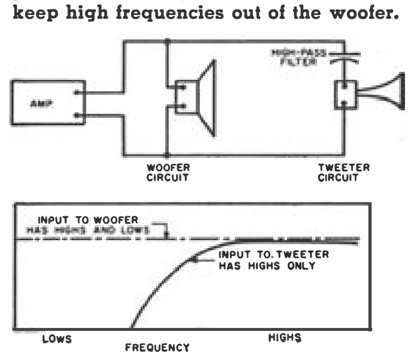


Fig. 1. Lack of network wastes power.

Fig. 2. Simple high-pass filter in tweeter circuit lets only highs into tweeter and provides tweeter protection, but does not keep high frequencies out of the woofer.



the over-all system; then we shall progressively add network elements to the system and observe their effect upon performance.

In Fig. 1 we have connected a woofer and a tweeter directly across a 10-watt amplifier. Assuming that the amplifier is a good high-fidelity type, we may then expect that it will have full-frequency-range output. Under this condition, the full-frequency range will be fed *equally to both the woofer and tweeter*. If both speakers are of the same impedance, the woofer will get half the power and the tweeter will get the other half. But in each case, the woofer and the tweeter will both receive the same full-frequency range. Under this condition, half of the high-frequency power available from the amplifier will appear at the voice-coil terminals of the woofer. But, being a woofer, it will not be able to reproduce these high frequencies. Consequently, all the high-frequency power that is fed to the woofer (half of the total high-frequency power) is entirely wasted.

At the tweeter terminals we find a similar condition of power available but with different results. Since half of the amplifier high-frequency power has already been lost in the woofer, the tweeter already has two strikes against it. Only half of the high-frequency power from the amplifier is available to the speaker to be reproduced as useful sound. Of equal importance is the fact that half the amplifier's *low*-frequency power also finds itself at the tweeter terminals. Naturally this represents a waste of half the total low-frequency power that would normally go to the woofer. Since the tweeter cannot reproduce the low frequencies, then the lows that find themselves at the tweeter are a total loss as far as reproduction is concerned.

Of equal importance is the fact that the tweeter itself may become, physically, a total loss under these conditions. Tweeters are invariably small and delicately made so that the last drop of efficiency may be extracted from the feeble high-frequency signals. Tweeters are just not built to handle heavy low-frequency signals that should normally go to the woofer, either from a power handling capacity or from a diaphragm excursion standpoint. It might not take more than a few moments of good, loud playing of a system without a network, such as is shown in Fig. 1, to destroy the tweeter. Thus, even if one wanted to start a system in its simplest form, the use of some type of network is absolutely essential if only as far as tweeter life expectancy is concerned.

High-Pass Filter

The simplest way to protect the tweeter against low frequencies is by inserting a capacitor in its circuit—a procedure which would normally block low frequencies. In this case, as in Fig. 2, we get two effects for the price of one. Destructive low frequencies are kept out of the tweeter; and a *high-pass* filter effect is obtained. Actually, it is this high-frequency passband effect that prevents the transference of the low frequencies into the tweeter. The passband of the capacitor may be chosen to coincide with the actual high-frequency output of the tweeter itself so that only those frequencies that the tweeter will eventually reproduce will get into the tweeter. The effect of this simplest type of "network" is the conservation of all the low-frequency power for utilization by the woofer and protection for the tweeter against damaging low-frequency power. It should be noticed, however, that in this simple system, since there is nothing in the circuit to prevent high-frequency power from getting into the woofer, that *half of the available high-frequency power is still lost in the woofer.*

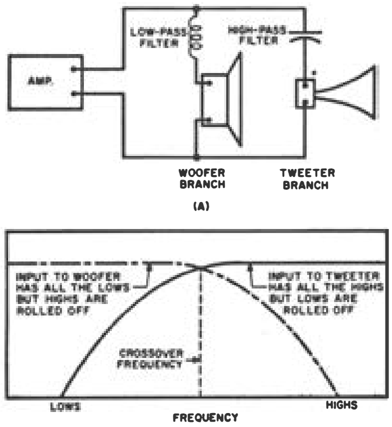
It would be worthwhile to digress briefly at this point to discuss the effects of such a "high-pass network" on the listening results obtained with this two-way system. The degree of effectiveness of this type of system insofar as the high frequencies are concerned will be greatest when the main speaker is *poor* in high-frequency response. If the main speaker is truly a woofer, then despite all the high frequencies that are sent into it, it won't reproduce any high-frequency sound. Consequently, when the tweeter and its high-pass element are subsequently connected across the amplifier, highs will begin to emerge from the system only as a result of the tweeter being connected and there will be a distinct audible difference. On the other hand, there are many cases where a tweeter and a high-pass element are connected, in the fashion just described, to a main speaker which is of the "wide-range" class. This situation will arise where one has originally installed a single speaker system for good over-all reproduction and then, at some later date, decides to build "up" from it to a multi-speaker system. Since the original *single* speaker installation is generally a good wide-range unit, it will reproduce high frequencies with fair efficiency and output. If a tweeter and a simple high-pass element are now connected across this type of speaker, the high frequencies will again split between the main speaker and the tweeter. *Half of the high-frequency power will still be reasonably reproduced by the original speaker* while the other half will go to the tweeter. Thus the increased over-all audible effect of adding the tweeter to a speaker which already reproduces highs will not have quite the impact as when connected to a woofer.

Full Two-Way Network

To overcome such a condition and to provide true *network* performance that will not only properly channel the various frequencies, but will improve over-all cleanness of reproduction, we have to add a single element to the circuit of Fig. 2. Fig. 3 shows a full two-way system with both high- and low-frequency controlling elements: a capacitor in the high-frequency channel to block the low frequencies from the tweeter and a choke in the low-frequency circuit to keep the high frequencies from the woofer. The capacitor is a high-pass element and the choke is a low-pass element. The combination of the two, usually referred to as an *LC* network, provides an electrical crossover function apportioning all the low-frequency power to the *woofer* and all the high-frequency power to the *tweeter.*

The audible effects of this sort of combination will be readily apparent. In almost all cases there will be fairly clean separation between the bands of sound radiating from the tweeter and the woofer. Where the efficiency levels of the two speakers are of the same order, the output sound from the tweeter and the woofer at the *crossover frequency* will be equal. Above the crossover point the output of the tweeter will be dependent solely upon the performance characteristics of the tweeter itself. However, as far as the woofer is concerned, its output above the crossover point does not simply fall away. See Fig. 4. It will drop off in a manner determined *first* by the output characteristic of the woofer, or main speaker itself, as discussed in the previous paragraphs. Then it will be further attenuated by the "roll-off" characteristic of the high-frequency limiting element (the choke) in the woofer circuit. The converse of this situation will hold for those frequencies below the crossover point. The woofer output will now be determined entirely by the woofer performance characteristic while the tweeter performance will be controlled first by the output characteristic of the tweeter and then additionally modified by how the high-pass frequency element in its circuit rolls off the low frequencies. As shown in Fig. 4, *the end result of the speaker network combination is a function neither of the network nor the speakers, but is controlled by both the electrical characteristics of the network and the acoustic output of the speakers.*

Fig. 3. Two-way network channels lows and highs to woofer and tweeter respectively.



Intermodulation Distortion

How does a two-way network of this *LC* type provide improved audio performance beyond the simple unit of Fig. 1 without the network? First, complete audio power utilization from the amplifier is now feasible. If there are a full 10 watts of low frequencies available from the amplifier, they will all go to the woofer and be reproduced there without half of them being wasted in the tweeter. Alternately, when there are a full 10 watts of high frequencies available from the amplifier, they will all go to the tweeter and be reproduced without half of them being wasted in the woofer. Then, of course, there will be full protection for the tweeter against damaging low frequencies.

However, most important from a performance standpoint is the fact that the full two-way crossover network system will provide considerable improvement (reduction) of the intermodulation distortion of the system. With the elimination of the highs from the woofer and their being channeled instead to the tweeter, these high frequencies are no longer bounced around by the large excursions of the woofer diaphragm which would be the case if both highs and lows were to come from the main speaker. By thus providing a separate tweeter diaphragm entirely independent of the more violent excursions of the large woofer diaphragm, considerable reduction in intermodulation distortion is possible, resulting in over-all cleaner sound.

Spatial and Level Response

When frequency division of this sort is practiced it is possible to overcome another defect of single-speaker operation, namely high-frequency beaming. In any large speaker, such as a typical 12" or, especially, a 15" unit the high frequencies tend to concentrate in a rather sharp beam in front of the speaker and high-frequency response over a wide listening angle is thus deteriorated. When, however, the highs are not reproduced by the large cone but are instead reproduced by a separate branch, then wide-angle dispersion of the high frequencies may be obtained, either through the use of dispersing type horn tweeters or a bent array of cone-type tweeters. Level control of the treble frequencies may also be easily accomplished now that they have a channel of their own. Such controls are referred to as "brilliance" controls. They raise or lower the entire *plateau* of the tweeter band, thus maintaining the full-frequency range of the tweeter despite the over-all output level of the unit.

Three-Way System

The principles of the two-way network may be readily extended to the popular three-way system. Such a system comprises a woofer, a mid-range unit, and a tweeter—speakers which reproduce, correspondingly, the low frequencies, the middle frequencies, and the treble frequencies. The same general attributes that were found for the two-way network system are now applicable to the three-way system but with more definition of detail. Obviously, with three-band operation, the separation of the high frequencies from the lows is more efficiently accomplished and more readily audible. Where in the two-way system, for instance, a crossover of 2000 cps may have been chosen, in the three-way system an upper crossover of 5000 cps may be utilized. Those frequencies from 5000 cps down to perhaps 350 cps would be carried by the mid-range unit, while below that all the low frequencies would come from the woofer. With this sort of separation, there is no question at all as to which band is carried by the woofer and which by the tweeter. Audibly, the difference between the woofer cutting off at 350 cps and the tweeter starting at 5000 cps is as clear-cut as night and day. The mid-range unit, bridging these two extremes, has a characteristic personality all its own, again very distinct from the other two branches.

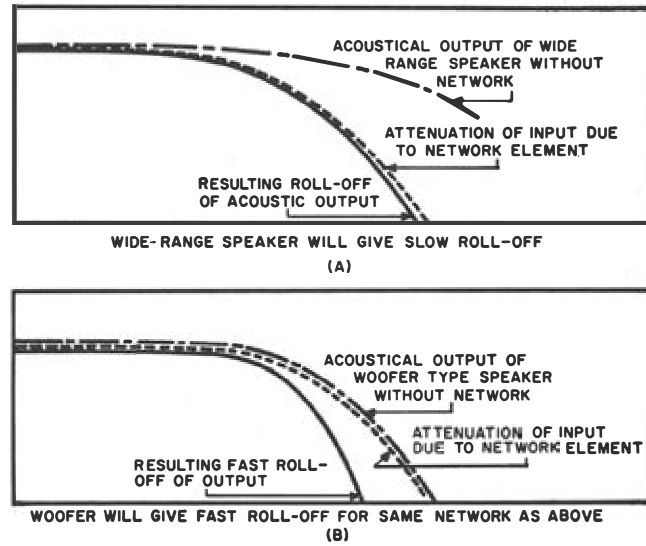


Fig. 4. The over-all system performance will be a function of both the performance of the loudspeaker and the network characteristic. Note the operation of the system with a wide-range speaker (shown at A) and with a woofer (shown at B).

Balance in Three-Way System

Psychological use is made of the mid-range tonal quality by referring to it as "presence." There are many who feel that reducing the level of the mid-range unit makes the performer recede somewhat in the background, while raising the mid-range level brings him forward—or increases his "presence." This controlling feature of mid-range "presence," along with the treble "brilliance" control, obviously makes the three-way system more versatile than the two-way set up.

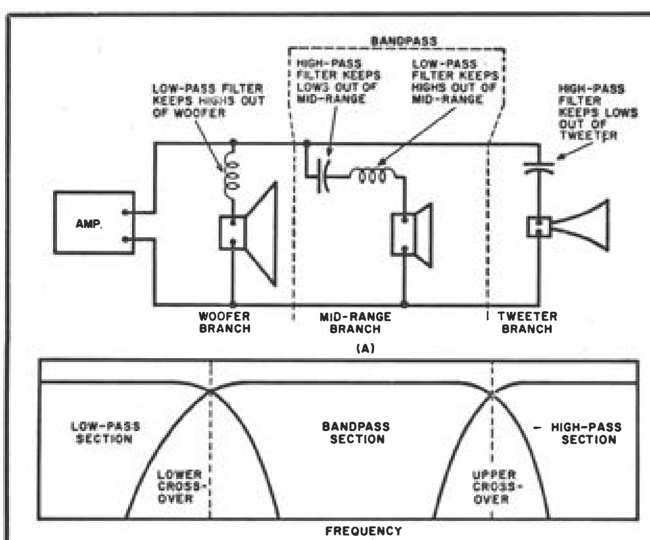
Since there is greater separation of the frequency bands in the three-way system than in the two-way system, we should expect further reduction of intermodulation distortion.

Mid-Frequency Controls

The filtering elements for a three-way network are actually a combination of the principles used in the design of two-way networks. The frequency controlling elements are, in the mid-range case, a capacitor to limit the low-frequency input to the speaker unit, in series with a choke to subsequently limit the high-frequency input to the same unit, as shown in Fig. 5. In this instance, the mid-range section is a *bandpass* type of filter, whereas the low-frequency section is a *low* pass and the high-frequency section is a *high* pass.

6 db and 12 db Networks

The electrical structures shown in Figs. 3 and 5 for the two-way and the three-way networks respectively are representative of the popular "6 db-per-octave attenuation" type. They are the simplest types to construct and provide an "easy" type of roll-off, that is, a gradual falling off of the frequencies. Where it is desired to provide sharper electrical attenuation of the signals going into the speakers, then a 12 db system may be designed by doubling up the filter elements—not two capacitors where one exists, but instead pairing off an inductance with an already existing capacitor, and a capacitor with an existing inductance. (The initial value of these *pairs* will not be the same for the 12 db as for the 6 db network. This will be discussed in detail in Part 2.) As an illustration of the conversion of a 6 db to a 12 db network, let us again consider the case of the two-way system of Fig. 3 which is a 6 db-per-octave network, the slow type, and examine the tweeter section. The capacitor passes high frequencies to the tweeter, as determined by the normally decreasing impedance of the capacitor as the frequency is increased. There is no sharp "yes and no" line of demarcation where the capacitor passes and where it does not pass power on to the tweeter. The manner in which this power is transferred to the tweeter is a



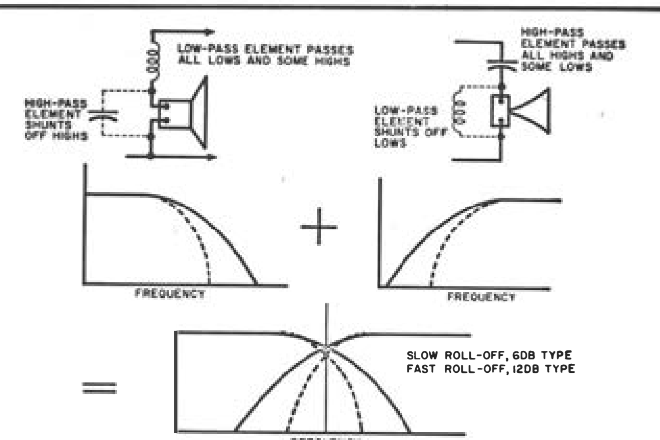


Fig. 6. Slow roll-off network may be changed to fast roll-off by shunting the speakers with frequency bypassing elements.

Fig. 5. Mid-range section is made by limiting lows with high-pass element, then limiting highs by means of low-pass element.

gradual one determined by the value of the capacitor in series with the tweeter. Some low frequencies below the crossover point will inevitably be transferred to the tweeter.

We can, however, *bypass* these low frequencies from the tweeter by shunting it with a choke, a *low-pass* element which has low impedance at low frequencies. Thus, if we have a tweeter in series with a capacitor, we may put *a choke across* the tweeter for faster roll-off of the low frequencies from the tweeter terminals, as shown in Fig. 6. As the frequency goes up, however, so will the impedance of the choke and the high frequencies that get passed on to the tweeter through the capacitor will not be bypassed from the tweeter by the choke.

In a similar manner, if a capacitor were to be put across the woofer terminals, it would bypass any high frequencies that might get through the choke to the woofer terminals and the high-frequency roll-off at the woofer would thus be faster than that provided by the choke itself. At the low frequencies the capacitor shunted across the woofer terminal would be relatively ineffective. It is thus possible to design a "fast" 12 db-per-octave network from a "slow" 6 db-per-octave network by simply adding, across each loudspeaker unit, a reactance element opposite in nature to that normally found in that particular branch of the circuit as a 6 db-per-octave network (but altered in value). Fig. 7 shows 12 db-per-octave counterparts of the simpler 6 db-per-octave networks.

We have intentionally avoided reference to numerical values for these reactive elements because it was desirable to first establish a general speaking acquaintance with the principles behind these common network systems. We have indicated why they were necessary; we have briefly discussed how they function; and the general nature of the differences among the various types. We are now ready to assign values to the components used in the various filter elements.

Part 2. Constructing the Network

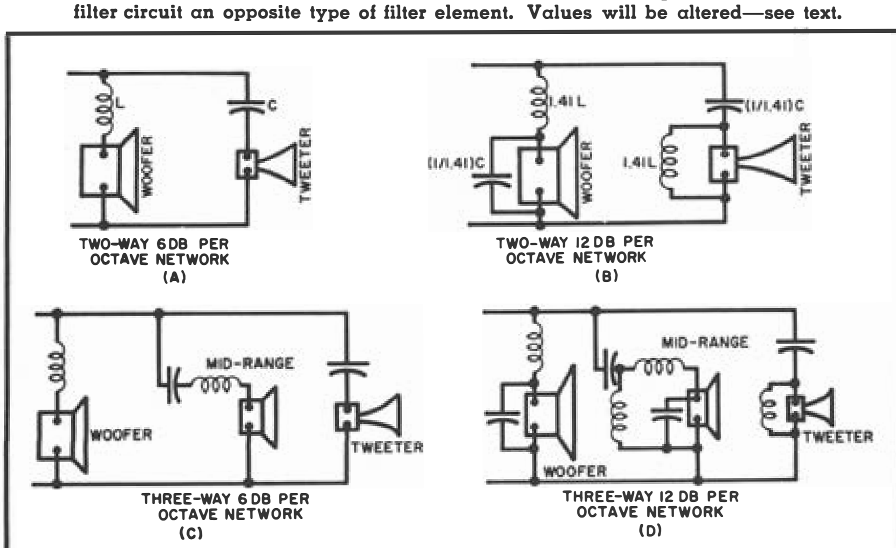
How to design and put together home-built precision networks for 2- and 3-way systems.

Now we must deal a little more specifically with this matter of the 6 db *versus* 12 db roll-off characteristic of the network to determine the source of the particular values chosen. These are not arbitrarily selected values—they are specifically related to the values of the choke or capacitor element that will provide a given crossover point for a given impedance.

Voltage Division at Crossover

By definition, the crossover point is that frequency where the drooping output of the low-frequency branch of the network crosses over the rising characteristic of the high-frequency branch (as indicated in Fig. 3 of Part 1). For a 6 db-per-octave network, the value of the capacitor in the tweeter branch and the value of the choke in the woofer branch are chosen to provide an a.c. impedance across those two respective elements, at the crossover frequency, which will be equal to the speaker impedance. Fig. 8 shows a simplified circuit of a two-way network with a low-frequency branch and a high-frequency branch, both tied across a common voltage source. Let us assume that the speakers are both 8-ohm units and that it is desired to design a network to cross over at 2000 cps. We will have to find an inductor and a capacitor that will each present an impedance of 8 ohms at this frequency. Having found such components (more details later on actually finding these) and inserting them in the network of Fig. 8A, we see that the voltage in the low-frequency branch has been equally divided across the choke and the woofer. In a similar manner, the voltage across the high-frequency branch has been divided equally between the capacitor and the tweeter. Consequently, the voltage across the woofer is equal to voltage across the tweeter. This is the crossover point where the drooping low-frequency characteristic crosses the rising low-frequency characteristic.

Fig. 7. A 6 db network is transformed to a 12 db network by adding into each filter circuit an opposite type of filter element. Values will be altered—see text.



6 db/Octave Attenuation

Now we come to the matter of the octave rate of attenuation of these drooping and rising characteristics. Consider, first, the drooping woofer branch characteristic of Fig. 8A. If it a crossover frequency of 2000 cps the inductance in the woofer circuit is equivalent to 8 ohms, then at 4000 cps (one octave higher) this inductance will present a 16-ohm impedance since the impedance is directly proportional to the frequency. When this 16-ohm impedance is now considered in series with the 8-ohm woofer, Fig. 8B, then the voltage across the choke becomes twice that across the woofer. On a db basis ($db = 20 \log E_2/E_1$) a 2 to 1 voltage ratio becomes 6 db. Thus, *after the crossover point,* the voltage drop-off across the woofer progresses at a rate of 6 db-per-octave. Thus, if we go up another octave the choke impedance doubles again, going from 16 ohms to 32 ohms, while the woofer still remains 8 ohms. The voltage in the woofer branch, Fig. 8C, is now at a 4 to 1 ratio. On a db basis, a voltage ratio of 4 to 1 represents a *total* drop of 12 db or, again, 6 db over the previous octave.

The same analysis may be applied to the tweeter branch and it may be shown, in identical fashion, that the tweeter circuit capacitor, when necessarily chosen to be equal in impedance *at the crossover frequency* to the tweeter impedance will, *below* the crossover point, continue to roll-off at the gradual rate of 6 db-per-octave. So it is seen that the *automatic 6 db rate* of this type of network arises from the simple necessity of choosing reactive elements in the two branches to divide the voltages equally across the various elements in the circuit so that the individual speaker terminal voltages will be the same at the crossover frequency.

12 db/Octave Attenuation

In Part 1 we discussed the general method of pairing off a capacitor with an inductance in each speaker circuit to convert a 6 db-per-octave network into a 12 db system. While "pairing off" is the general procedure, the values to be used in converting from a 6 db to a 12 db network need some modification. Thus, if a choke had been originally selected to have an impedance of 8 ohms at the crossover frequency (and equal to the speaker impedance), then it would have to be multiplied by a factor of 1.41 when the systems were

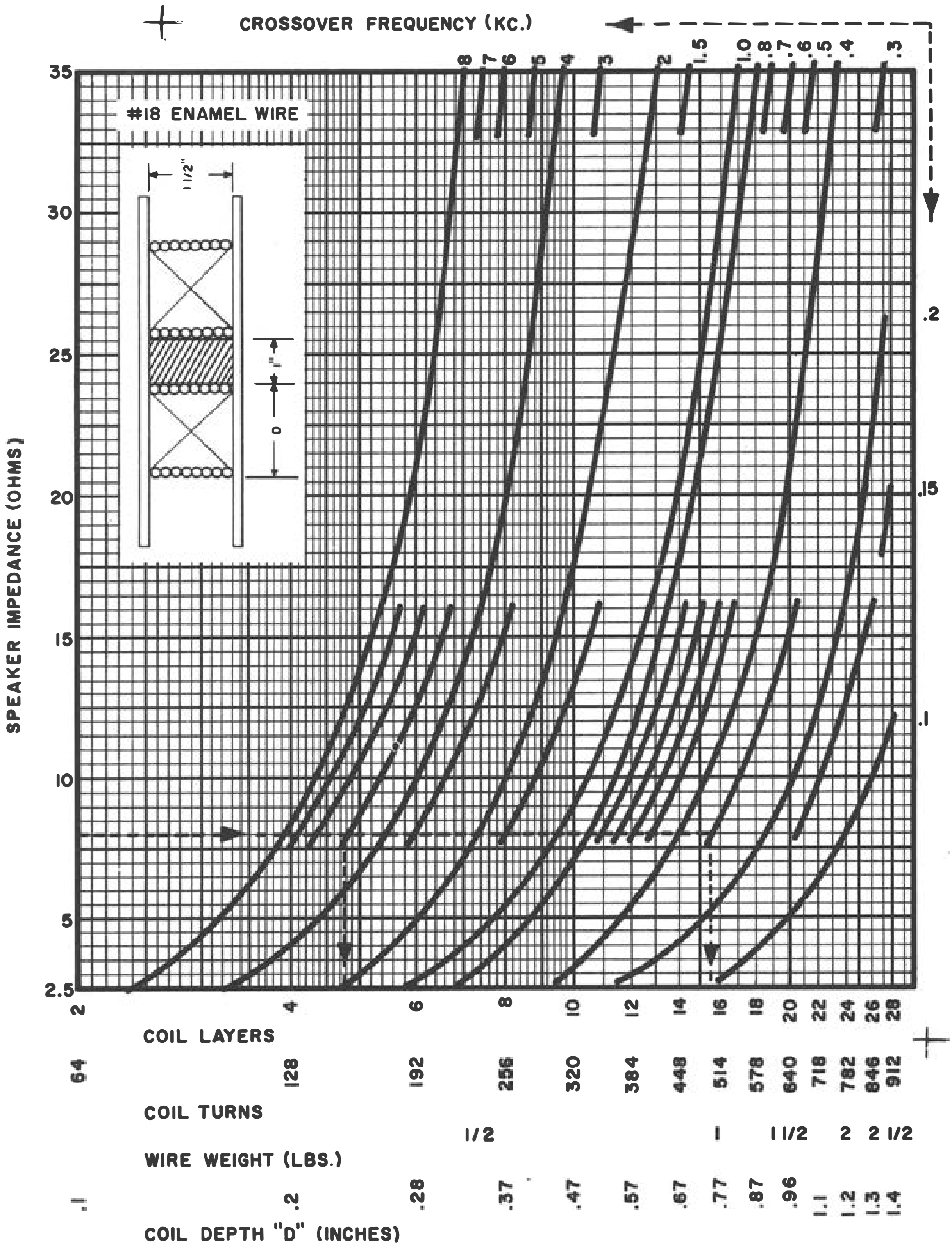


Chart 1 (Above). A master chart giving all physical winding data for crossover network chokes based on speaker impedance and desired crossover frequency. Choose the speaker impedance on the vertical scale, move over horizontally to the curve which represents the desired crossover frequency, and then move down to the horizontal scale which gives all the vital statistics of the coil needed for the chosen conditions for a 6 db per octave network, shown in circuit (A) to right. To use chart for 12 db per octave network, circuit (B), simply multiply speaker impedance by 1.41 and proceed as above.

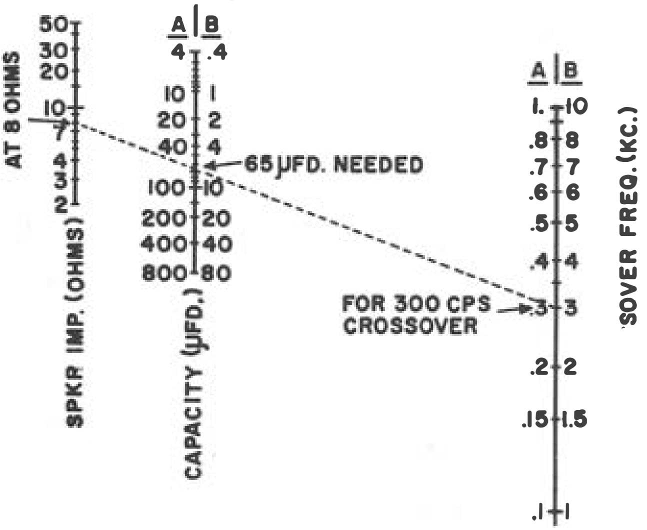
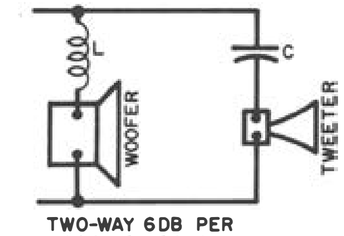
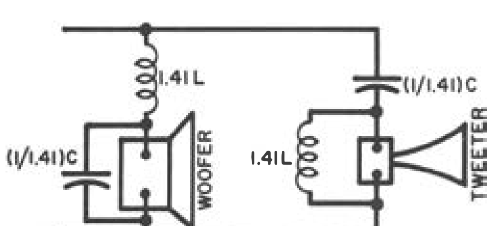


Chart 2 (Left). Auxiliary chart to be used in conjunction with Chart 1 for determining the value of capacity needed to cross over at a given speaker impedance. To use chart, simply lay a straight edge between the point on the first column representing the impedance of the speaker and the point on the last column representing the crossover frequency desired. The point where the straight edge crosses the center column is the value of capacity required. Be sure to use the two columns marked "A" together, or the two columns marked "B" together. This chart is to be used for a 6 db per octave network only. In order to obtain capacity values for a 12 db per octave network it is only necessary to divide the value of capacity obtained by 1.41. For the lower values of capacitance, oil-filled capacitors are preferred. For the higher values, electrolytics are employed.

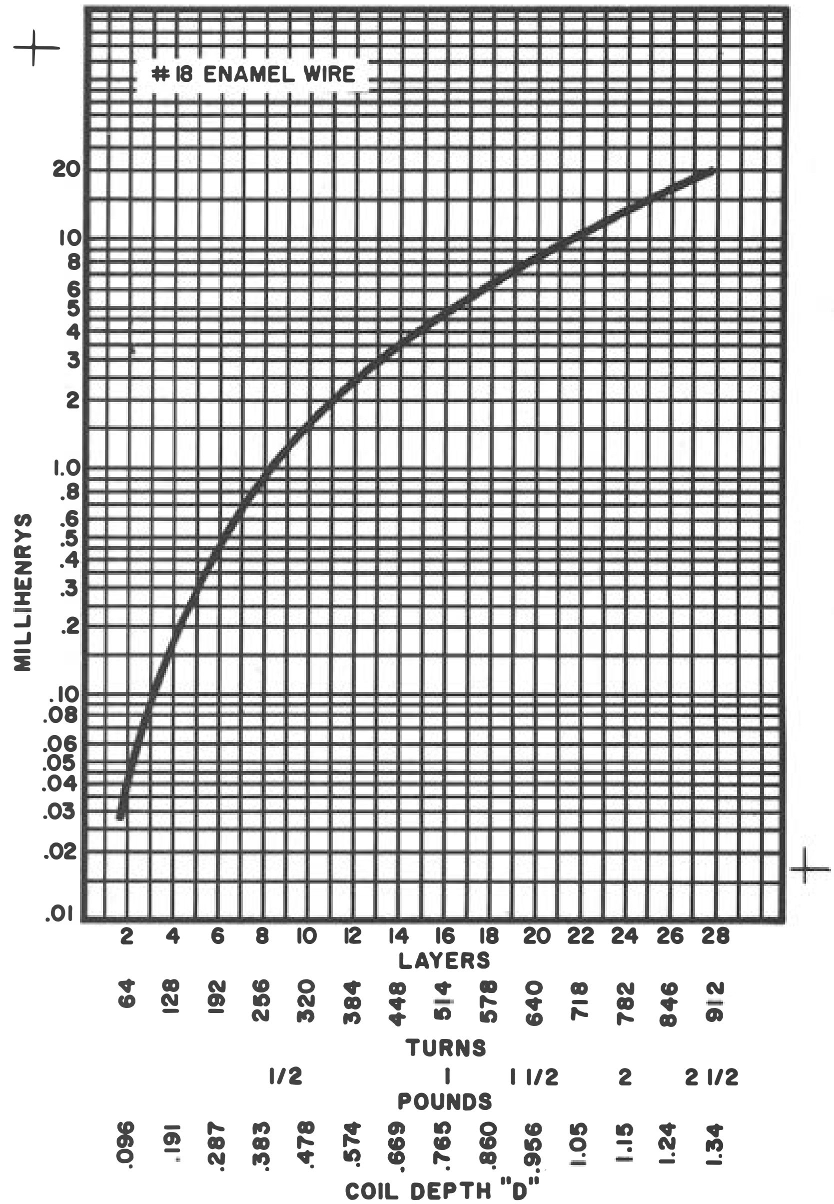
Chart 3 (Below). The actual inductance values of network chokes used in the master Chart 1 are given here. This chart makes it possible to wind coils of given inductance if the construction details shown along the bottom axis and in the drawing at the center of this page are followed. Inductance values from about .03 up to 20 millihenrys are covered here.



TWO-WAY 6 DB PER OCTAVE NETWORK (A)



TWO-WAY 12 DB PER OCTAVE NETWORK (B)



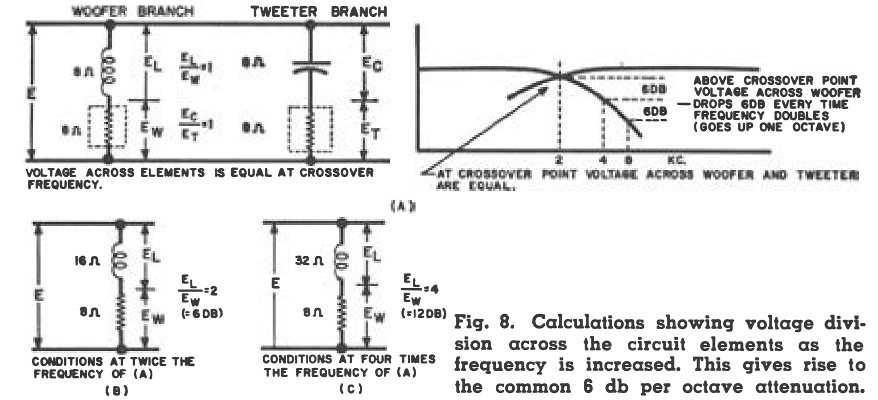


Fig. 8. Calculations showing voltage division across the circuit elements as the frequency is increased. This gives rise to the common 6 db per octave attenuation.

changed to a 12 db-per-octave network as indicated in Fig. 7 of Part 1. Similarly, the capacitance of the component which had been originally chosen for the tweeter circuit of the 6 db-per-octave network would have to be divided by a factor of 1.41 when the conversion is made. Once these new values have been determined by modifying the 6 db values, they may then be paired off to provide the 12 db network. Calculations similar to those in Fig. 8 may be made when using these revised values to plot out the network branch voltage which will drop at the rate of 12 db-per-octave *after the crossover point when going in either direction.*

It may seem that we put the "cart before the horse" in giving details on how to convert from a 6 db to a 12 db-per-octave network before we had discussed how to select the simple values for the 6 db network. However, since we had treated such conversion last month as part of the general philosophy of network design, it was deemed logical to carry over that discussion in terms of "numbers" so that a transition might be made to the problem of selecting real values of inductances and capacitances for a particular network.

Building the Coil

Although Chart 1 provides all of the practical details for making the coil for any given impedance and for any given crossover frequency, we have included another chart for the purist who still wants to know the inductance of his coil. If one were truly ambitious, he could wind one master coil with several taps along the depth for experimental purposes. Charts 1 and 3 give the actual curves of an experimentally checked master coil wound of #18 enamel wire on the coil form shown on the chart page. Along the abscissa are four scales: first, the number of layers, then the number of turns, then the pounds of wire that are necessary for a given inductance, and finally the coil depth. The coil form is made with a 1" wooden dowel as the core and the end pieces of hard ¼" Masonite. A series of ⅛" holes were drilled along a radius of one of these end pieces so taps could be brought out anywhere along the depth of the coil.

Chart 1 gives the details of the coil configuration for any desired frequency and speaker impedance. Choose the speaker impedance on the vertical scale, move over horizontally to the curve which represents the desired crossover frequency, and then move down to the horizontal scale which gives *all* the vital statistics on the coil for the conditions selected for a 6 db-per-octave network.

To use Chart 1 for 12 db-per-octave networks multiply the value of the speaker impedance by 1.41 and proceed as above. This, in effect, increases the inductance value by 1.41 times, a requirement for a 12 db-per-octave network.

The corresponding capacity to go along with the chosen inductance is easily determined. One may make a simple calculation of capacity by using the formula: $C = 1/2\pi f X_c$ where f is the crossover frequency and X_c represents the reactance of the capacitor chosen to be equal to the speaker impedance at the crossover frequency. C will be the capacity required for the tweeter branch. Alternately, Chart 2 may be used to pick off the actual capacitor value for a given impedance at a given frequency. Here, again, as in the case of the coil, the value found for the capacitor is for a 6 db-per-octave network. For a 12 db-per-octave crossover, divide the capacitance value obtained by 1.41.

Typical Three-Way Network Parts

The very important matter of the type of capacitor to use deserves individual treatment, but consideration of this point will be deferred to the last so that we may illustrate the actual selection of component values for a typical three-way system. Let us assume an 8-ohm system with a crossover at 300 cps between the woofer and the mid-range and an upper crossover at 5000 cps between the mid-range and the tweeter. This system was shown in Fig. 7 of Part 1. The choke for the woofer is selected from Chart 1 by coming in from the 8-ohm point (speaker impedance) on the vertical scale to the 300 cps curve and then down to the horizontal scale where it is indicated that very nearly 16 layers (or 500 turns) of wire will be required on the coil form, that just under one pound of wire will be needed, and the coil depth will be approximately ¾". This is all the information required for winding this woofer circuit coil.

Now, the low-frequency blocking capacitor of the mid-range circuit will have to be equivalent in impedance to the speaker at the 300 cps crossover frequency. From Chart 2 the value of this capacitor turns out to be 65 µfd.

Moving to the upper crossover frequency of 5000 cps, the high-frequency limiting choke in the mid-range circuit should have an impedance of 8 ohms at this frequency. Again from Chart 1, we select 8 ohms on the vertical scale, move horizontally to the curve representing 5000 cps, then vertically down the horizontal scale where we find that the coil will consist of 5 layers of wire (160 turns), will utilize approximately ¼ pound of wire, and will be about ¼" thick. The corresponding tweeter branch capacitor at this crossover frequency point will also have to have an impedance of 8 ohms and from Chart 2 this turns out to be 4.2 µfd. (call it 4). Thus all the details for winding the coils and choosing the right capacitor values are readily available if you know the speaker impedances and the desired crossover frequencies.

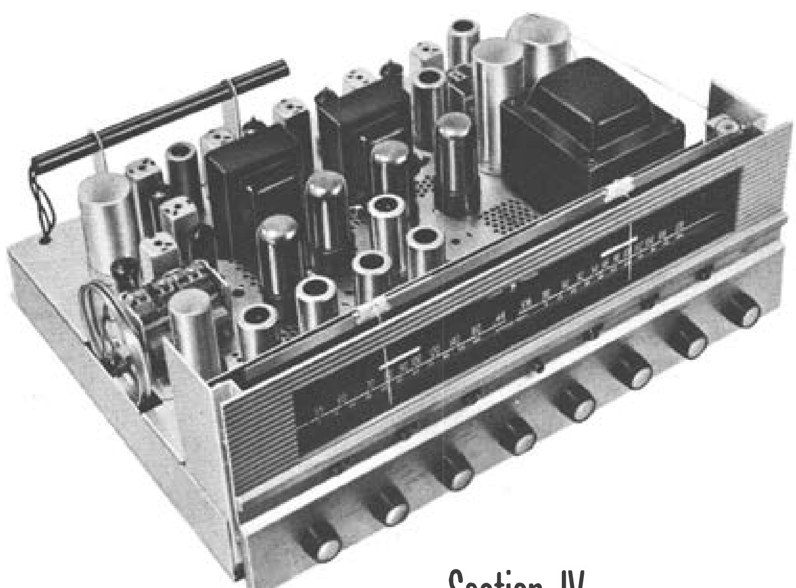
To convert this network into the 12 db system shown in Fig. 7D, Part 1, the inductance values of the chokes should be multiplied by 1.41 and the capacities divided by 1.41 and then paired off as previously described.

Type of Capacitors

We must now discuss the controversial question of the type of capacitor to be used in audio dividing networks. It has been generally conceded that one can't go wrong if he uses good oil-filled or paper capacitors. However, there is the matter of cost for such units. A 60 µfd. capacitor, even one rated at comparatively low voltage, may not fit one's pocketbook as well as it does the network data. This problem has been overcome in commercial equipment by using non-polarized electrolytic types where large capacities are required. These are comparatively cheap but they do have their shortcomings. In practice it has been found that the actual capacity of a batch of electrolytics, all rated the same but measured at the higher frequencies, may vary by as much as 25 to 30% from the rated value. In some instances it has also been found that the impedance of the non-polarized electrolytic may climb at the very high frequencies causing a tweeter loss.

This latter loss may be easily overcome by shunting the electrolytic with a small paper capacitor, 1 µfd., for example, which will serve to keep the impedance of the capacitor section of the tweeter branch low at the high frequencies.

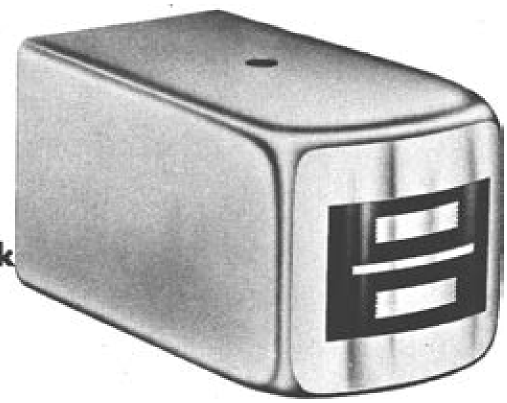
Looking prejudice squarely in the eye, there would seem to be no reason as yet for not using electrolytics, polarized or non-polarized, for network construction.



Section IV

FM and Tape Recorders

Here is what some of the very newest extra-small-gap heads can do with quarter-track tape when operated at speeds of only 3.75 inches/sec.



Here are examples of new quarter-track record/playback (left) and erase (below) tape recorder heads.

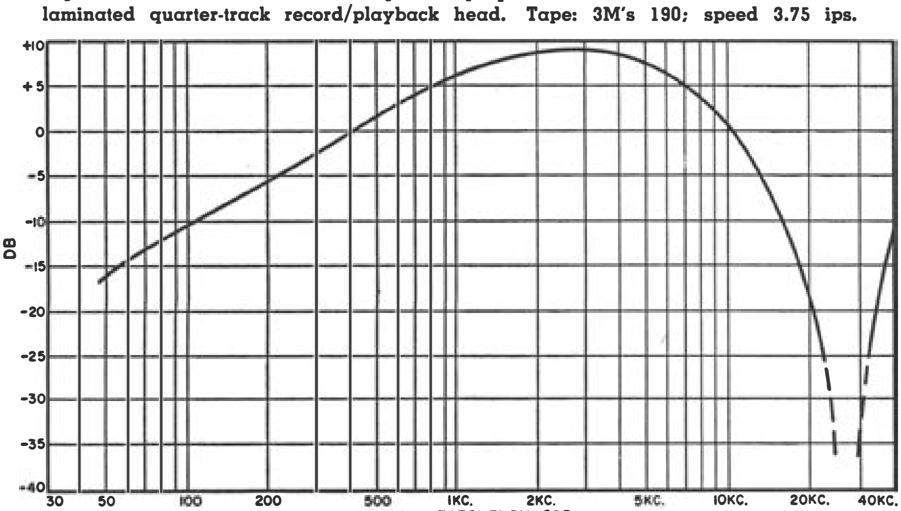
Slow-Speed Tape Recording

By **JOHN W. HOGAN**

WHEN two large tape recorder manufacturers released specifications on a four-track, 3.75-ips system last year, the impact was sufficient to cause a complete re-appraisal of the medium by other members of the tape recording industry including equipment manufacturers, dealers, distributors, tape-music suppliers, and, of course, audiophiles and other ultimate users of tape recorders. As of this writing, several machine manufacturers are supplying reel-type machines that will operate at 3.75 ips and incorporate "quarter-track" heads for this type tape. In a few cases, recorder manufacturers have provided a vertical head mechanism that enables the quarter-track head to be shifted to a center position for playback of the older half-track stereo tapes. As an alternate plan, other manufacturers are supplying both half-track and quarter-track heads on the same machine.

Performance evaluation of quarter-track, 3.75-ips operation is a difficult task. In direct comparison with stereo discs, the new tape system offers advantages common to any tape unit. These include: negligible quality deterioration with extensive playing, professional cross-talk rejection characteristics, excellent dynamic range, and good frequency response characteristics. Recent LP discs are capable of high-fidelity performance—particularly when new and in good condition. Unfortunately, the basic noise level and corresponding usable dynamic range may deteriorate so that some audiophiles "tape" their new stereo discs and preserve the original. Channel cross-talk rejection on the finest stereo disc systems is only 20 to 30 db while even the poorest tape system betters this figure by at least 10 db.

Fig. 1. Constant-current-record, unequalized-playback characteristics for Model TLD-L laminated quarter-track record/playback head. Tape: 3M's 190; speed 3.75 ips.



Compared with 7.5-ips, half-track tape, the new tape system requires careful analysis. In the first place, only a few manufacturers of 7.5-ips recorders have taken advantage of the full potential of the basic tape recording art. Most machines use single laminar heads which are characterized by excessive core losses and subsequent reduction in dynamic range. Amplifiers, in many cases, are not properly equalized and, generally, the inherent tape noise level—which should determine the noise in the basic system—is exceeded by amplifier hum and noise. In the author's opinion, a well-designed, quarter-track, 3.75-ips machine using laminated heads, properly adjusted bias and equalization characteristics, and precision drive components is capable of providing equal or superior performance to many of the recently manufactured 7.5-ips machines.

The high retail prices of recorded music tapes have dampened the enthusiasm of many audiophiles and music lovers. Adoption of the lower speed and the narrower track widths will provide one answer to this cost problem and, in all probability, open the tape recording field to the popular music market as well.

Technical Considerations

Keeping in mind these factors, let's

consider a few of the technical problems involved in the design and manufacture of components for tape recorders in the medium-priced field. We will discuss record/playback and erase heads, amplifiers, as well as integrated systems for quarter-track operation. Record/playback head design, production, and limitations will be covered in some detail since this component is of such basic importance in any system designed for high-fidelity performance. In other words, no amount of engineering ingenuity or production "gimmicks" can offset the liabilities of inadequate head performance in a tape recorder.

The Heads

A basic description of a magnetic recording head is simple. It consists of a closed core assembly with a precisely lapped and spaced gap over which the tape passes during the recording and/or playback process. The head also includes a winding assembly consisting of multiple turns of copper wire to energize the head on recording and provide the induced voltage on playback.

When the head is used exclusively for playback, the effective gap should be small enough so that it doesn't approach the wavelength of the highest frequency on the tape. Fig. 1 shows a constant-current-record, unequalized-playback curve of a laminated quarter-track head at 3.75-ips. The effective gap coincides with the null reading at 28 kc. and is, therefore, 3.75/28000 or .000134 inch. The actual or mechanical gap on a head of this type, as checked under a microscope, measured .000090 to .0001 inch. Discrepancies between effective and mechanical gap figures are the result of production faults, such as improper finishing, which might result in the situation diagrammed in Fig. 2. Fortunately, in this case, as tape/head wear progresses, the effective gap will begin to approach the mechanical gap dimensions. For reliable playback performance the first frequency null, occurring when the wavelength on the tape equals the effective gap, will be twice the highest frequency desired. Thus, Fig. 2 represents a head well suited to 14 kc. playback performance with a quarter-track system at 3.75 ips.

In actual playback operation, the highly permeable structure of the head offers a relatively easy path for flux from the tape section that spans the gap. Tape motion across the gap causes the varying magnetic strength of the recorded flux signal to be transferred through the core and thereby generate a voltage in the associated winding as:

$$e = -N\frac{d\phi}{dt}$$

Since the induced voltage amplitude is proportional to frequency, a constant-flux recording will result in a uniformly rising output of 6 db or two times per octave. Two basic types of losses will completely alter the picture. The first of these, wavelength or gap losses, has already been briefly discussed. Here, as the physical wavelength of the recorded flux signal gradually approaches the effective gap, the induced voltage will approach a minimum or zero value, depending on the parallelism of the two gap surfaces.

The second group of losses are classified as electrical or frequency losses. They include core losses, such as eddy current and hysteresis losses, and copper losses in the winding. Other related factors that cannot be readily defined as losses and yet affect performance include capacitance, resonance effects, etc. Fig. 3 shows the composite effects of these various losses at 3.75-ips speeds.

Curve 1 in Fig. 3 shows a 6 db-per-octave curve. If there were no losses during recording or playback, a constant-current (flux) recording would produce this type of response from an open-circuit playback head into a voltmeter. Curve 2 includes nominal losses introduced during the recording process. These will be described later in more detail. Curve 3 shows the effect of electrical or frequency losses on playback over and above those sustained in recording. Curve 4 shows the resultant of all these losses plus playback loss from gap or wavelength effects and is the same curve that was obtained experimentally in Fig. 1. Other factors which enter into playback performance include: tape-to-head contact, dirt, misalignment, etc.

A magnetic recording head designed specifically for recording applications would differ from a playback head in two respects. First, it would have a larger gap length to allow the medium to be subjected to several bias cycles. This eases the bias and record cur-

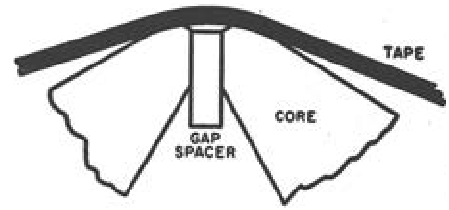


Fig. 2. Enlarged drawing showing how improper finishing will produce a large difference between mechanical and effective gap.

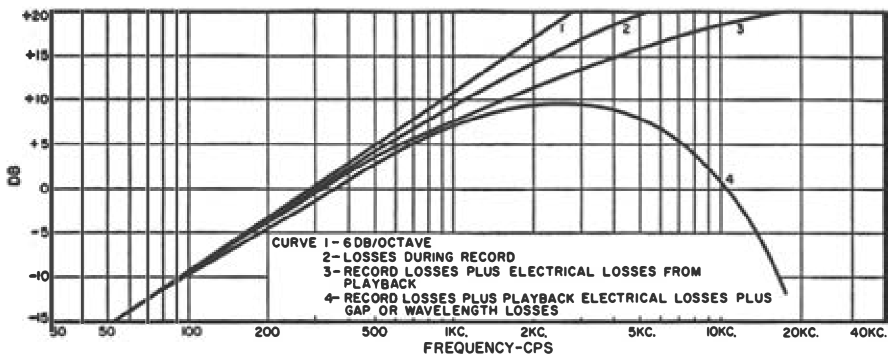


Fig. 3. Effect of losses during record and playback to show how resultant constant-current-record, unequalized-playback response differs from 6 db per octave rise.

Fig. 4. Curve showing how the signal current is combined with the bias current.

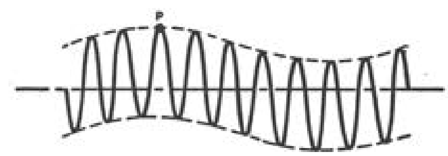


Fig. 5. Curve showing particles of the medium as they are carried through first an increasing field due to the tape entering the record head gap (points S, 2, 3, and 4) and then through a decreasing field when the tape leaves the record head gap (points 5, 6, and F). Point F represents the remaining magnetization.

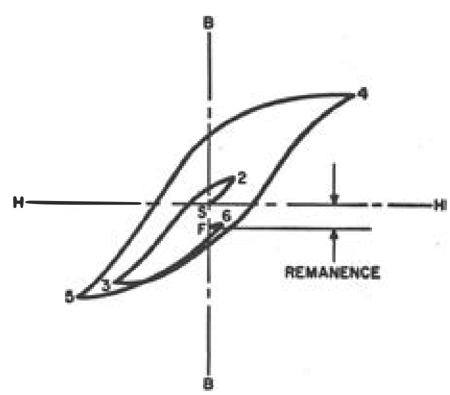
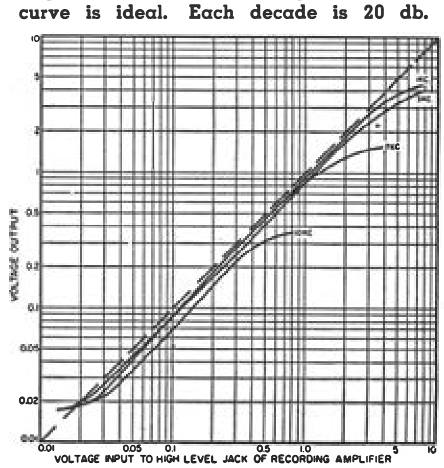


Fig. 6. Equalized input-output, record/playback curves for a quarter-track, 3.75 ips system at various frequencies. The record and playback equalization and bias settings are as shown in Fig. 8. Dashed curve is ideal. Each decade is 20 db.



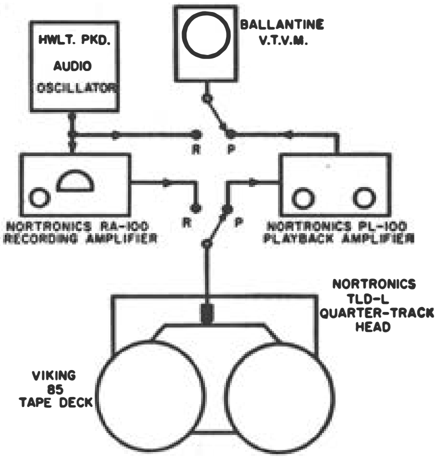


Fig. 7. Test setup employed for Fig. 6.

rent drive requirements and further allows a relatively thick conductive spacer material to deflect the bias and signal fields toward the medium.[1] The second general design requirement of a head to be used exclusively for recording it that the relative number of turns or head electrical impedance be low enough to reduce winding capacity losses; this is especially true when the higher bias frequencies are used. Professional tape recorder manufacturers will often use a core material for a record head that has higher flux handling capabilities than the high-nickel alloys used for playback heads.

The a.c. bias-record operation is complicated and a thorough explanation of the principles involved would require more space than can be spared here. The basic point to remember is that the bias and signal fields are added together in the recording operation. This is shown in Fig. 4. Considering a given point in time as P and remembering that several bias cycles occur as the tape passes across the gap, the action of entry and exit of tape coinciding with time point P would be to expose the tape particles to first an increasing intensity bias field (the signal may be considered as a d.c. component effectively deflecting the bias field upward at the point P) and then a decreasing field as the particles under consideration leave the gap area. As the particles are carried through this series of increasing and decreasing hysteresis loops, the ultimate remanence is represented by F in Fig. 5.

Any discussion of recording techniques requires consideration of bias frequency and amplitude. The bias frequency should be at least four or five times the highest audio frequency desired. Reducing the frequency below this figure will generate beat frequencies that will ultimately result in audible distortion when any amplifier-circuit non-linearity is encountered. The amplitude of the bias signal current is extremely important in critical tape recording applications. It assumes even greater importance at low speeds because demagnetization due to over-biasing will have a relatively greater effect on high-frequency performance. Using too little bias will improve the relative high-frequency response but at the expense of signal-to-noise ratio and distortion. At 3.75 ips the correct compromise for achieving good frequency response, low distortion, and wide dynamic range or good signal-to-noise characteristics is a bias slightly below that figure which will give maximum response at 1 kc. well below the 3 percent distortion level.

As previously mentioned, losses occur during recording which contribute to a non-uniform magnetization level on the tape if the head is energized with a uniform magnetizing current. These losses fall into two groups: tape losses which cannot be compensated when the tape is magnetized to its upper limit; and electrical or frequency losses and bias demagnetizing losses which can be counteracted irrespective of the recording level on the tape.

The discussion thus far has been confined to the single-purpose, single recording or playback head. Far more common, even on some semi-professional recorders, are the combination record/playback heads. These heads are usually "weighted" toward the factors that make a good playback head. At 3.75 ips, however, even with effective gap lengths near .0001 inch, excellent recordings can be made.

Several manufacturers have conducted extensive experiments to determine optimum equalization, dynamic range, distortion and frequency response characteristics for 3.75-ips operation. Their results have shown that precision in quarter-track record/playback head manufacture is such that frequency response at 3.75 ips is comparable to that obtained with present-day single laminar half-track heads at 7.5 ips. In addition to frequency response, however, the importance of usable linear dynamic range at any specific frequency in the audio band should not be overlooked. The family of curves of Fig. 6 shows maximum usable linearity ranges at 1 kc., 3 kc., 7 kc., and 10 kc. at 3.75 ips. with a *Nortronics* TLD-L laminated quarter-track head in the setup of Fig. 7.

Typical NAB-type equalization curves for the 3.75-ips system with a *Nortronics* TLD-L laminated head are shown in Fig. 8. Curve 1 represents the constant-current-record, unequalized-playback curve of the head with a bias chosen as previously outlined. Curve 2 represents the record-current equalization. This curve, in conjunction with Curve 1, will produce the unequalized playback response of Curve 3. Curve 4 represents the playback equalization necessary for the complete system.

It is interesting to note that the TLD-L laminated narrow-gap, quarter-track head has approximately the same response at 3.75-ips as the older TLD-S single laminar half-track head has at 7.5 ips. There are two reasons for this: the gap length of the TLD-L head is about one-half the gap length of the TLD-S and the core losses are considerably less, as shown in Fig. 9.

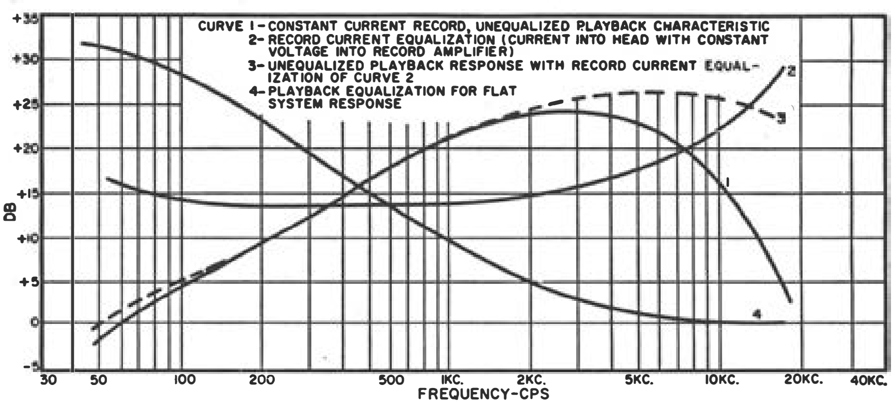
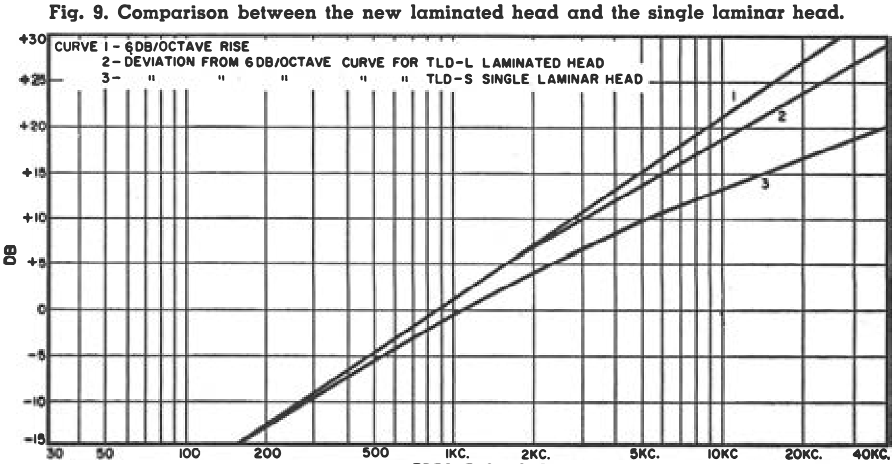
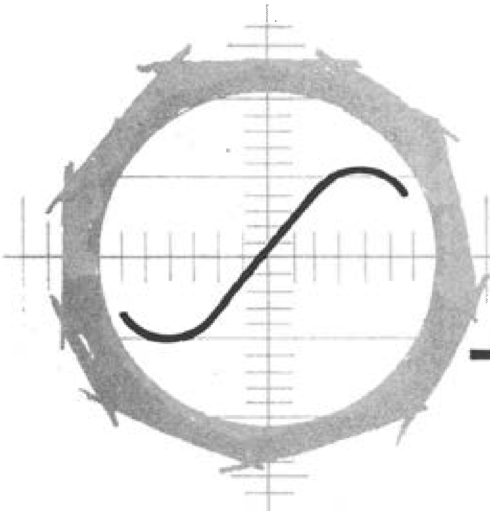


Fig. 8. Typical NAB-type equalization curves for laminated quarter-track head.

Fig. 9. Comparison between the new laminated head and the single laminar head.





By **JOHN M. DOYLE**

Tips on FM Alignment

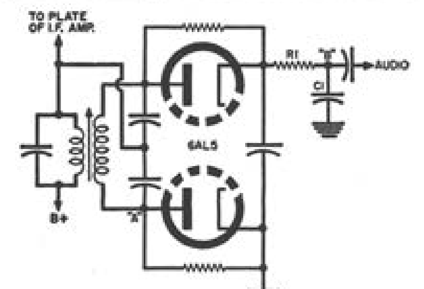
Circuits using various types of detectors each have their own bugaboos. Practical techniques.

SINCE MUCH has been written about the alignment of FM receivers, there is not much point in simply reviewing well-known procedures. Therefore, specific problems peculiar to certain types of FM circuits will be our main concern. It will be convenient to concentrate on the various detector circuits, but consideration will be given to related alignment procedures. Considered in order will be discriminator-type detectors, ratio detectors (balanced and unbalanced), locked-in oscillator detectors, and gated-beam detectors.

Before any test equipment is connected to the tuner, the latter should be adjusted to a point where no signal is received. This is done because shorting out of the oscillator in an FM receiver cannot usually be accomplished reliably. Since the inductance of the jumper used is likely to approach that of the oscillator coil, oscillator operation tends to continue, although at a different frequency.

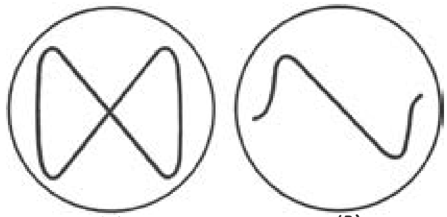
For the initial phases of alignment, it is often more practical to forego a sweep generator in favor of a more conventional r.f. generator—one that has been set to the i.f. with 400-cps amplitude modulation. The latter type of signal will be found to provide good accuracy and convenience, especially in indicating exact center frequency during detector alignment. The audio signal also comes in handy for such other purposes as checking AM rejection and detecting oscillation by its effect on the receiver's audio output.

Fig. 1. A version of the discriminator.



Connection of the signal-generator leads to the receiver chassis in the vicinity of the detector or the final i.f. stage frequently results in the development of standing waves. As a result of these, oscillation may be encountered when i.f. adjustments are attempted. To eliminate this effect, generator leads often have to be re-dressed carefully.

Refer now to Fig. 1, which is the schematic for one version of the discriminator-type detector. The primary of the discriminator transformer is adjusted first, for maximum output. Ordinarily the indicating instrument is connected from the center tap of the



(A) (B)

Fig. 2. Swept detector response with scope synced at (A) 120 and (B) 60 cps.

secondary to ground, across the load resistor of the secondary. In most versions of the discriminator, this is easy to locate. In this case, the v.t.v.m. is connected from point "A" to ground.

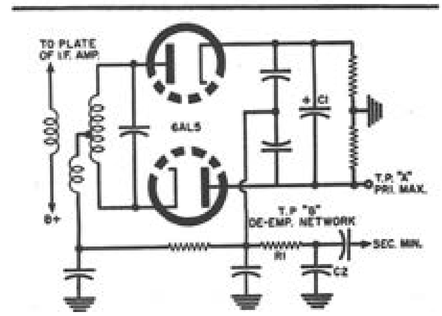
The meter is then transferred to the output of the de-emphasis network (point "B") for the adjustment of the discriminator-transformer secondary. The desired reading here is zero d.c. volts. As the proper setting of the secondary is reached, the meter pointer drops from a definite amount of deflection to zero almost instantaneously. Also the audio modulation from the generator will almost completely disappear at the same point. We therefore have two indicators, one visual and the other aural, for accurately determining the proper point.

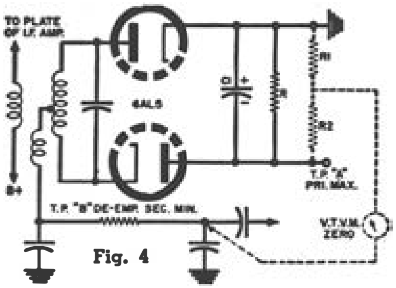
If the v.t.v.m. has a zero-center scale, this will be convenient in making the adjustment. A slight movement of the adjustment in one direction will then result in a negative reading, while over-adjustment in the other direction will result in a quick swing through zero to a positive reading.

For final alignment, the sweep generator is brought into play. While a discussion of the conventional setting of generator controls is not intended here, it should certainly be pointed out that sweep width should be increased to the maximum that is consistent with convenient observation of the trace. If sweep is not wide enough, there are several difficulties that can mask the proper alignment point. To mention some, misalignment of the i.f. channel, regeneration, spurious response from the tuner, and standing waves can all obscure the scope pattern to some extent. When the sweep is, for example, 450 kc. wide, the trace on the screen will show the whole response curve and indicate such troubles. If the sweep is too narrow, we may also end up by setting the crossover point on one side of the i.f. response curve.

Occasionally the connection of an oscilloscope to the receiver may result in pickup of external signals, noise, or hum voltages. Internal regeneration or oscillation may also result. These conditions cause modulation or distortion of the observed pattern. To guard against these annoyances, the lead to

Fig. 3. Ratio detector, balanced type.





SHUNT CONNECTION OF MATCHED PAIR RESISTORS AND CONNECTION OF V.T.V.M. INDICATED BY DOTTED LINES

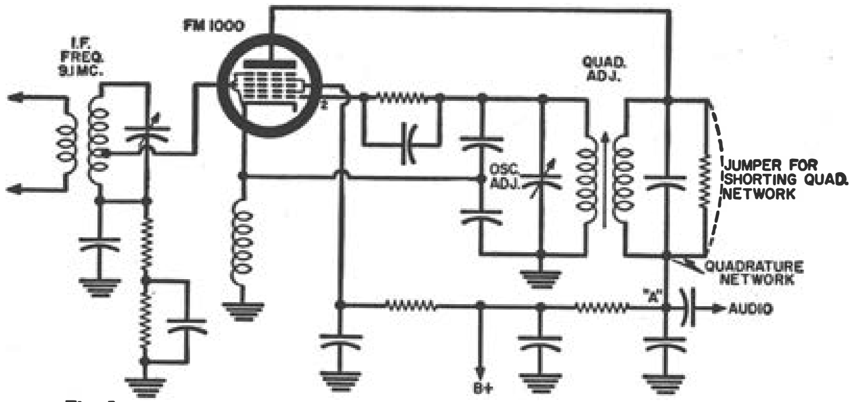


Fig. 5

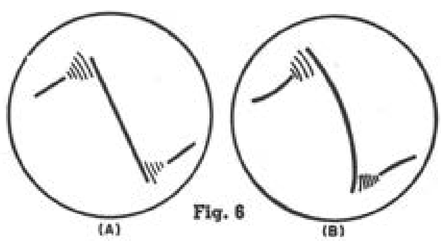


Fig. 4 (Upper left). One version of the unbalanced ratio detector.

Fig. 5 (Above). Locked-in oscillator-detector found in Philco sets.

Fig. 6. Correct (A) and poor (B) response for oscillator-detector.

the vertical-input terminals should be kept as short as possible and well shielded. If these troubles persist, a resistor in the range between 50,000 and 100,000 ohms may be placed in series with the vertical-input lead of the oscilloscope. In the circuit of Fig. 1, for example, this is necessary to avoid detuning. Keep this resistor as close as possible to the point of contact in the receiver.

A few words are necessary here concerning the final alignment of the i.f. section. When adjusting the frequency setting of the sweep generator, always make certain that the response curve displayed is for the proper intermediate frequency rather than for a harmonic. It is easier to make a mistake of this kind than one might expect, especially if one happens to be working with an instrument that is not completely familiar. However, a simple check eliminates this possibility of error.

With the response curve showing on the screen of the scope, move the receiver tuning control back and forth. If the frequency setting of the generator is correct, the pattern will remain stationary; if not, the pattern will move off the screen as the tuning knob of the receiver is turned.

The reason for this movement lies in the design of those sweep generators where this difficulty is likely to occur. The FM is imposed on an oscillator of fixed frequency, usually in the range between 25 and 60 mc. This signal then beats against another variable r.f. oscillator, providing sweep around the desired frequency by heterodyne action. However, more than one heterodyne output is produced.

For example, assume that the fixed swept oscillator is operating around 45 mc. and that it is beating with a variable oscillator adjusted to 55.7 mc. By adding these two we get 100.7 mc. By subtraction we get 10.7 mc., the desired signal. In addition, the second harmonic of the 45-mc. output is at 90 mc. Thus, there are at least three sweep signals available in this case to which circuits in the receiver may respond, although only one of them is desirable.

If the trace observed on the oscilloscope is the result of either the 90-mc. or the 100.7-mc. signal, it will move off the screen as the receiver's tuning control is rotated because these signals beat with the local oscillator in order to enter the i.f. section. On the other hand, if it is the true 10.7-mc. output, it will stay put. Furthermore, this does not take into account such other adverse effects as image response.

Assuming that i.f. alignment is correct, final oscilloscope alignment of the discriminator is performed by first connecting the scope to point "A" of Fig. 1, as already noted, for the primary adjustment, in which maximum amplitude of the response curve is sought. The scope lead is then moved to the de-emphasis network (point "B") to obtain the crossover pattern, which is adjusted for maximum symmetry, particularly at the crossover point. The sweep frequency of the scope may be set to 120 cps, in which case the pattern should resemble that of Fig. 2A. If the scope is synced at 60 cps, the pattern of Fig. 2B will be obtained.

We will now consider the ratio detector, both balanced and unbalanced types. Information previously given for signal-generator settings and attachment of instrument leads applies equally well here. We first connect the v.t.v.m. across electrolytic capacitor C_1 (point "A" in both Figs. 3 and 4) and ground to adjust the primary of the detector transformer for maximum curve amplitude and to make the i.f. adjustments as well. In the balanced type, we next connect the v.t.v.m. to the de-emphasis network, between point "B" (R_1, C_2 in Fig. 3) and ground, and then adjust the secondary for the rapid zero reading previously described.

To make this adjustment in the unbalanced type, a pair of high-value, matched resistors—100,000 ohms or more—is usually shunted across R in Fig. 4. Connection of the meter is then made between the usual point in the de-emphasis network and the junction of these two temporarily added resistors. To avoid this bothersome procedure, the secondary may be adjusted by ear for minimum audio as previously described, using a conventional r.f. generator with a 400-cycle amplitude-modulated signal. This time saver is

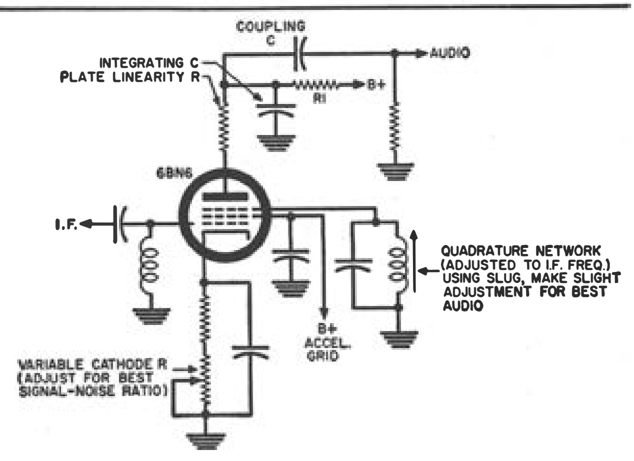


Fig. 7. A gated-beam detector using the 6BN6 tube. The two adjustments in this circuit are the slug in the quadrature tank and the AM rejection control in the 6BN6 cathode.

particularly acceptable if final alignment will be completed with the oscilloscope anyhow.

When the FM generator and scope are brought into play, remember that we rely on the latter to indicate amplitude changes during adjustment of the detector primary and the i.f. channel. Therefore limiting action in the detector is disabled temporarily by disconnecting electrolytic capacitor C_1 in Figs. 3 and 4 and making the scope connection at this same point. In the balanced circuit, it is satisfactory to remove either end of the capacitor. In the unbalanced version, the negative terminal should be removed.

When i.f. adjustments have been completed, reconnect C_1 and transfer the scope to the de-emphasis network, as already noted, to adjust the transformer secondary for maximum symmetry of the crossover pattern. In some sets, it will be necessary to re-adjust the primary to a very small degree at this time to get the crossover point to straighten up.

The locked-in oscillator-detector employing the FM1000 tube was developed by *Philco.* Refer to Fig. 5. The oscillator grid, pin 2, is first grounded which causes the section to operate as an AM detector. The i.f. frequency used here is 9.1 mc. The output indicating instrument is connected either to point "A" or to any other audio-frequency signal point. Alignment of the i.f. section is then made by employing the same procedure as given for the discriminator detector. When the i.f. adjustments have been completed, the ground jumper on the oscillator grid is removed, and the quadrature circuit is short-circuited, as shown in the schematic, to disable the locking action of the plate feedback into the oscillator section. With the signal generator set for 400-cycle AM output at 9.1 mc., adjust the oscillator trimmer until a beat note is heard in the loudspeaker. This audio note results from the heterodyning of the oscillator and i.f. signals. Continue to adjust the oscillator trimmer until a condition of zero beat is obtained (a null between two low-pitched growls).

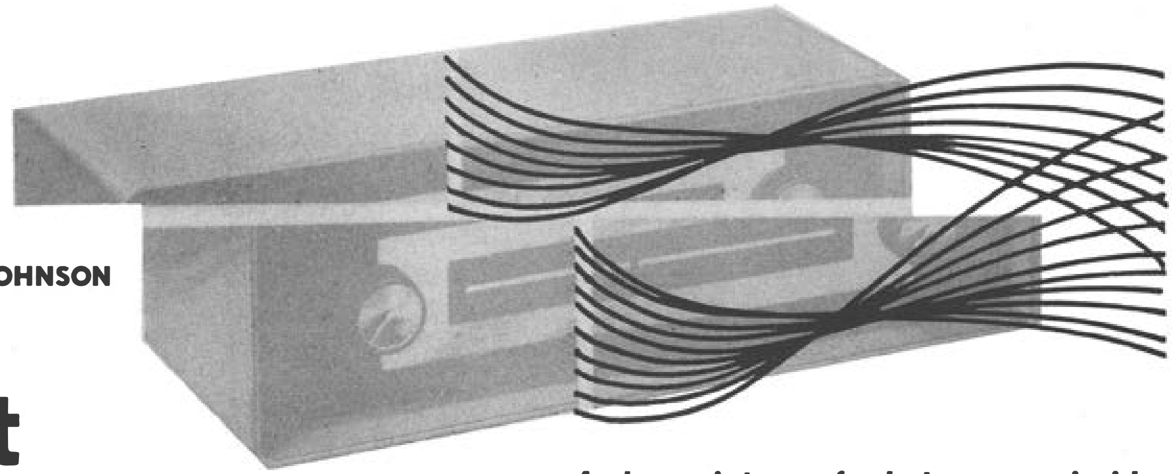
When the short circuit is removed from the quadrature circuit, the audio signal will again be heard. Reduce the signal generator output to the lowest value at which a usable signal can be heard. Re-adjust the tuning slug in the quadrature circuit until a zero beat is again obtained. Repeat the entire operation as necessary until the removal of the short from the quadrature circuit has minimum effect on the zero-beat condition obtained when the oscillator trimmer is adjusted. When the jumper is removed from the quadrature circuit, the complete response curve can be observed on the screen of the scope with FM sweep. The quadrature slug is then adjusted for maximum linearity. Fig. 6A shows a proper response curve, while that of Fig. 6B is poor.

Once the detector is properly aligned, the oscillator grid is again grounded and the receiver front end adjusted in the usual manner.

In receivers employing the 6BN6 type of gated-beam detector, Fig. 7, alignment of the i.f. stages is accomplished by measuring the r.f. voltage at the signal grid of the detector. A suitable probe is therefore required with the indicating instrument.

Attach the r.f. probe to the detector signal grid and adjust the i.f. transformer for maximum r.f. voltage reading. Next, tune in a local FM station for maximum indication and adjust the quadrature circuit coil for best audio results. Finally, tune in a weak signal accompanied by a large amount of noise and adjust the variable control in the cathode circuit for the best signal-to-noise ratio. It may be necessary to disconnect the antenna and do some juggling with the degree of coupling to the receiver input in order to maintain the noise level sufficiently high so that changes can be detected readily. A slight re-adjustment of the quadrature coil may then improve fidelity.

The main point to remember is that poor alignment can cover up an otherwise skillful repair job. When in doubt, go over all adjustments.



By
J. RICHARD JOHNSON

Drift & A.F.C. in FM

A clear picture of what goes on inside the set can speed up troubleshooting when oscillator stability is involved.

ONE OF THE ETERNAL problems with FM receivers is that of oscillator drift, although compensating capacitors and a.f.c. have gone a long way toward minimizing the trouble. While such measures make life more pleasant for the receiver owner in normal use, they can, when trouble develops, complicate things for the technician. In the popular AM-FM types, the interrelationship between circuits used for both modes of operation doesn't help matters either. Before tackling problems in FM stability, the technician should know some pertinent things about the receiver circuitry.

The detector in an FM receiver demodulates the i.f. signal. The center frequency of the i.f. signal must stay close to the center of the amplitude-frequency characteristic of the detector. One of the reasons is illustrated in Fig. 1. In each of the three diagrams, the frequency-modulated i.f. signal is shown on the lower, vertical axis in such a way that it can be projected to the detector characteristic shown above it. The a.f. signal that results from demodulation is shown along the horizontal axis to the right of the detector characteristic. With this arrangement, the input-signal values can be projected geometrically through the detector characteristic to the a.f. output signal waveform. In each of these diagrams, the input signal is assumed to be modulated by a sine wave.

Fig. 1A illustrates the condition of perfect tuning. The center frequency of the i.f. signal corresponds exactly to the center frequency of the detector characteristic. Within the limitations of a small, inevitable amount of curvature in the actual characteristic, the modulating sine wave is faithfully reproduced.

Fig. 1B illustrates what happens with a small amount of detuning. Because the positive excursions of the input signal now reach into the substantially-curved portion of the detector characteristic, one half of the a.f. signal is flattened, and the signal is thus noticeably distorted.

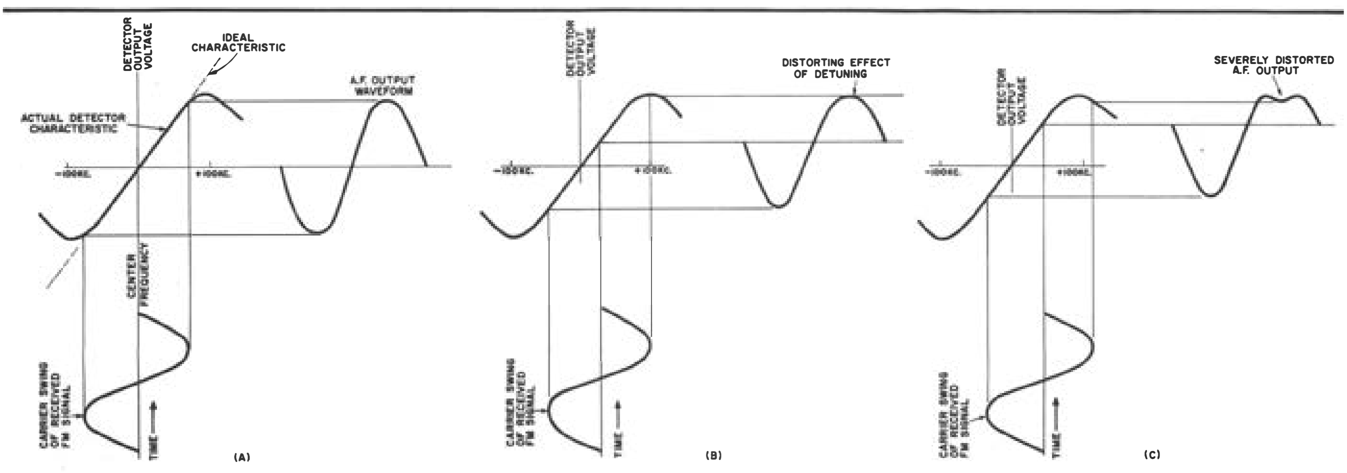
Fig. 1C illustrates what happens with a large amount of detuning. Positive excursions of the input signal extend beyond the edge of the detector characteristic. The result is extreme distortion, as shown.

The lower the amplitude of the modulation signal, the less the FM signal deviates from its center frequency, and the less likely it is to swing past the detector's linear portion. Thus, one characteristic of detuning distortion is that it gets worse as modulation-signal amplitude increases. In some cases, only the high-level peaks of the program material distort. Even when the detuning is small and the signal is within the so-called "linear" portion of the detector characteristic, the fact that slight non-linearities are not symmetrical about the received signal center frequency adds to distortion.

It thus becomes clear that, for full-quality FM reception, the tuning should be kept as nearly perfect as possible. The i.f. signal center frequency should be within a few kc. of the detector's center frequency.

The i.f. center frequency is determined by the local oscillator. Ordi-

Fig. 1. In properly tuned FM set (A) linear detection results in undistorted audio output. Detuning (B, C) distorts audio.



narily the i.f. is the difference between the oscillator frequency and the (lower) FM carrier being received. For example, with the receiver tuned to 90 mc., the oscillator should be at 100.7 mc. The difference, 10.7 mc., is the generally used i.f. for FM receivers. When the local oscillator in the front end drifts as little as 20 or 30 kc., the corresponding shift of the i.f. band can result in annoying distortion and noise interference. The problem of noise arises sometimes even before distortion is noticed, because detuning during drift reduces the signal amplitude to the limiter. In cases in which the signal is weak enough to be just on the threshold of adequate limiting, drift allows such noises as automobile ignition pulses to be serious obstacles to quiet listening.

Reasons for Instability

The local oscillators of FM receivers are much more susceptible to drift than their AM receiver counterparts. The main reason is that the values of inductance and capacitance used in FM circuits for tuning are very much lower than those in AM circuits. These reactances are so low in FM that they do not greatly exceed those of tube and stray circuit capacitances. The latter vary considerably during temperature changes and thus change the resonant frequency of the oscillator tuned circuit.

At the same time, the physical dimensions of the oscillator coil change due to thermal expansion. Since a temperature rise tends to increase the size of most components, and thus also increase the values of inductance and capacitance, the frequency of the oscillator tends to drift downward.

Of course the greatest drift occurs when the receiver is warming up from a "cold" start, as this is when the greatest temperature change takes place. Most FM receivers do tend to drift during the first 15 minutes or so after being turned on from a cold start. After that time, the oscillator should "settle down" and be relatively stable. However, if the oscillator is not carefully designed, or if there is a defect in its circuit, drift may continue indefinitely, causing distortion and excessive noise due to mistuning. It will also be subject to frequency change as a result of line voltage fluctuations.

Temperature Compensation

Some FM receivers use *negative-coefficient* capacitors to compensate for drift during the warmup period and for general stability. A negative-coefficient capacitor is one whose capacitance becomes *less* as its temperature increases, thus reversing the usual trend. If connected in a tuned circuit, such a capacitor causes the resonant frequency to become *higher* as temperature increases.

As previously noted, ordinary oscillator circuits tend to drift downward in resonant frequency during warmup, so the negative-coefficient capacitor tends to compensate for this drift. Such capacitors are manufactured in several standard ratings. The receiver designer attempts to choose the capacitance and coefficient that will most nearly cancel the normal uncompensated drift in the circuit.

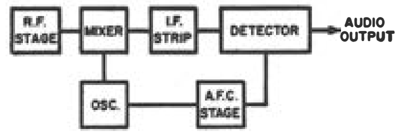


Fig. 2. How a.f.c. locks oscillator.

As an example of the ratings of negative-coefficient capacitors, one of the most popular coefficients is 750 parts-per-million per degree centigrade, designated as "N750." This means that this capacitor changes capacitance by 750 millionths of its value for each change of one degree centigrade in its temperature. The "N" stands for negative, indicating that the capacitance changes in a direction opposite to that of temperature change. In other words, if the temperature rises the capacitance goes down, and if the temperature falls the capacitance rises.

It is important that the technician be familiar with such data because, if a temperature-compensating capacitor is to be replaced, the replacement must be correct. Otherwise, bad oscillator drift may result.

A. F. C. Circuits

Another device that helps combat tendencies toward oscillator drift is the automatic frequency control (a.f.c.) circuit. This circuit probably came into use mainly to aid in providing less critical tuning to the receiver owner. However, in so doing, it overcomes the effects of oscillator drift within limits.

The basic principle of a.f.c. is illustrated in the block diagram of Fig. 2. The FM detector develops a d.c. control voltage. This voltage is normally zero when the front end is correctly tuned, but becomes plus or minus when the front end is off tune one way or the other. The control voltage is applied to the "a.f.c. tube," which has the more general name of "reactance tube." The reactance tube converts the variations of d.c. control voltage into variations of *reactance* across the oscillator tuned circuit.

The reactance thus applied to the oscillator becomes part of its tuned circuit, and helps to determine the resonant frequency. If, through temperature drift or line voltage variation, the oscillator should shift frequency so the receiver is out of tune, the detector control voltage changes the reactance exhibited by the reactance tube. This, in turn, changes the resonant frequency of the oscillator in such a way that this frequency shifts back toward what it should be for proper receiver tuning.

A basic reactance-tube is shown in Fig. 3. This circuit simulates and applies capacitance to the oscillator at terminals *A* and *B*. When an a.c. voltage is applied to a capacitor, the ensuing current *leads* the voltage by 90 degrees; this is a basic characteristic of a capacitor. The reactance-tube circuit does the same thing: when an a.c. voltage (such as an oscillator signal) is applied to terminals *A* and *B*, the resulting current leads the voltage 90 degrees. The oscillator tuned circuit, to which these terminals are connected, cannot tell the difference between the reactance-tube circuit and a capacitor. It thus reacts in the same way as though a real capacitance were present across *A* and *B*.

Now let's look into how the reactance tube sets up this voltage-current relationship:

1. The a.c. voltage from the oscillator is applied to points *A* and *B*, and thus also to *C* and *R* in series (Fig. 3).

2. The capacitance of *C* is deliberately made small enough so that its reactance in the circuit will be much higher than the resistance of *R*. Thus the series circuit composed of these two elements, consisting of a large reactance and a negligible resistance, is almost entirely capacitive. Therefore the alternating current that flows through this combination *leads* the voltage applied to it (at terminals *A* and *B*) by almost 90 degrees.

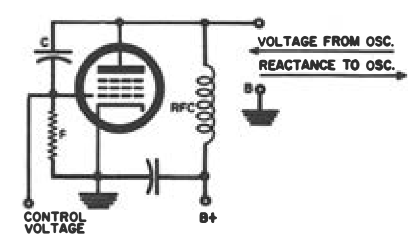
3. The voltage across a resistor is in phase with the current passing through it. Thus the *leading* alternating current applied to the resistor develops an in-phase voltage across this component. Furthermore, this new, developed voltage must then lead by 90 degrees the original a.c. voltage applied through terminals *A* and *B*.

4. Since *R* happens to be the grid resistor of the tube, we may now say that the a.c. voltage applied to the grid (voltage across *R*) leads the a.c. voltage at the plate (applied by the oscillator).

5. The plate *current* of a tube is directly controlled by and in phase with the signal *voltage* applied at the grid. Thus the alternating *current* at the plate of the reactance tube that results from the a.c. voltage at the *grid* also leads the a.c. *voltage at the plate* by 90 degrees.

6. With current leading voltage, the circuit has the properties of a capacitor. Since the output of the reactance tube (plate to ground) is connected across the oscillator's resonant circuit through terminals *A* and *B*, the reactance tube simulates a capacitor added to the oscillator tank to help in tuning it.

Fig. 3. Basic reactance-tube circuit.



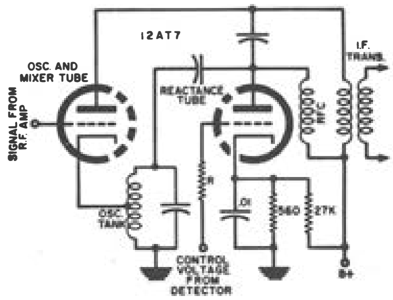


Fig. 4. A typical a.f.c. and oscillator circuit, using a twin triode.

It is worth mentioning that some reactance-tube circuits are designed to act as inductors. However, the capacitive type described here is the more usual case. Also, while we have established the nature of the circuit as capacitive, we have said nothing about the amount of capacitance that is simulated.

In order to readjust oscillator frequency as needed, we must be able to vary the amount of simulated capacitance used for retuning. At this point, we may consider the role of the control voltage also applied to the grid of the reactance tube in Fig. 3. The more positive this d.c. voltage becomes, the greater will be the plate current. Since more current will flow through a larger capacitor (less capacitive reactance) than through a smaller one, a relatively positive control voltage will increase the simulated capacitance applied to the oscillator tank. This will lower oscillator frequency.

Obviously, the circuit must be so wired that, when local-oscillator frequency drifts too high, the corresponding d.c. imbalance in the detector's output must be applied in the positive direction as a control voltage. Conversely, if the oscillator goes lower in frequency, control voltage from the detector becomes more negative. This reduces the reactance tube's plate current, decreasing the simulated output capacitance (increasing reactance), and oscillator frequency is tuned back up to where it should be.

In many modern FM tuner circuits, a triode is used as the reactance tube. This makes it convenient to use one section of a dual triode for this purpose while the other section is used for another function, usually that of oscillator.

One typical modern circuit is shown in Fig. 4. A dual triode (12AT7) is used for the oscillator, mixer, and a.f.c. (reactance) tube. No external capacitor is shown for the function of C (in Fig. 3). It happens that the triode's grid-to-plate capacitance is large enough to assume this role. Notice that quite a bit of positive voltage is applied to the cathode of the reactance tube through a 27,000-ohm resistor. This biasing prevents loading of the oscillator circuit from becoming excessive.

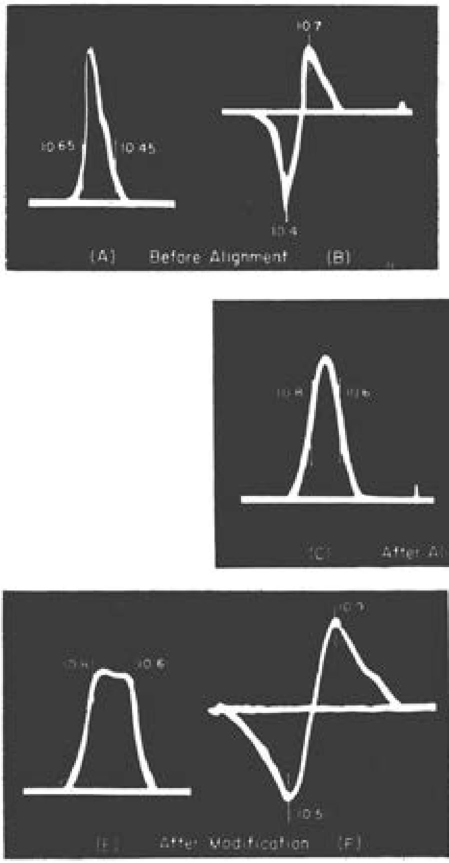


Fig. 1. The actual case record of one of the tuners modified by the author. Response curves (A), (C), and (E) were taken at the second limiter grid, while curves (B), (D), and (F) were at the discriminator.

Improving your FM Tuner

By **W. B. BERNARD**
Capt., USN

Some changes that will take only a few hours to make will pay large dividends in listening satisfaction.

THERE have been, and still are, some FM tuners, both kit type and factory assembled, which have insufficient i.f. bandwidth to give top fidelity reception of broadcast programs unless the received signal is an extremely strong one. It matters not what the sensitivity of the receiver is or how much AM rejection it has if it produces distortion on signals of reasonable strength.

The unsatisfactory i.f. bandpass situation that exists in some cases is due, principally, to two causes. The i.f. transformers may be under-coupled and there may be feedback from the output of the i.f. strip back to the input which will distort the i.f. response. Even if feedback is not a problem in the original circuit it may become one as transformer coupling is increased.

To make the tuner revisions described here it will be necessary to have a sweep-frequency oscillator, a marker oscillator to cover the i.f. range in the vicinity of 10.7 mc., and an oscilloscope. This equipment will permit visual alignment and checking of the results of the modification.

The actual sequence of operations will be determined by the type of transformers used and whether the receiver uses a limiter-discriminator or a ratio detector system. If ¾" x ¾" transformers are used, it is not practical to vary the magnetic coupling of the two windings so the additional coupling must be capacitive. Because this additional capacitive coupling will depend, to a great extent, on that already present, adjustments on this type transformer are most easily made when the transformers are checked in the tuner. In the case of some of the larger transformers which may be easily removed from their shields, the magnetic coupling is increased by moving one turn of one of the windings closer to the other winding, as shown in Fig. 3. This may be done with the transformer installed in the tuner or in a transformer test jig, the diagram for which is shown in Fig. 2.

This test jig is so wired that the signal is fed into an amplifier which drives the primary of the transformer under test. If the transformer is an interstage type, the secondary is connected to an infinite-impedance-type detector which will rectify the i.f. signal. The detector output is then fed into the oscilloscope to allow observation of the response curve. At the discretion of the experimenter, a third socket may be installed near the infinite-impedance detector socket. A 6AL5 may be used in this socket either as a discriminator detector or as a ratio detector. An FM test signal is, of course, required. The test jig is not essential if only one tuner is to be modified. If a number of i.f. transformers are to be wound or modified, the jig will save both time and effort.

Fig. 1 is an actual "case history" on one of the tuners modified by the author. The tuner in this case was a kit which, according to the instruction booklet, was furnished with pre-aligned transformers which required no further adjustment to provide moderately satisfactory reception. The tuner was assembled according to instructions and, upon completion, the i.f. response to the second limiter grid was checked. The result is the narrow, peaked curve shown in Fig. 1A. It can be seen that the ± 100 kc. points are down about 15 db from the peak. The discriminator curve for this condition is shown in Fig. 1B. The peaks are 300 kc. apart and the line connecting the peaks is far from being straight although the input signal was sufficient to provide considerable limiting. In fact, the limiting was sufficient to obscure the marker pip except when it was moved to the peaks of the curve.

In the original condition, acceptable reception could be obtained on two of the strongest signals but most of the other signals in the band were badly distorted. The next step in the process was to align the i.f. system. The resulting curve at the second limiter grid is shown in Fig. 1C. The center frequency has been moved to 10.7 mc. and the ± 100 kc. points are now about 5 db down. Fig. 1D shows the discriminator curve. It is still 300 kc. peak-to-peak but the line connecting the peaks is now much straighter. Listening tests showed that additional stations could be received satisfactorily but that some of the stations which were limiting adequately were giving distorted reception when the modulation percentage was high.

It should be pointed out that the curves of Figs. 1C and 1D are not at all unusual with either kits or factory assembled tuners. The i.f. response can be broadened by stagger-tuning the transformer windings but this results in a reduction of gain in the i.f. system which, in many cases, is not acceptable. The preferable remedy is to increase the coupling between the primary and the secondary of the interstage transformers. In the case described, the transformers were of the ¾″ x ¾″ type so the additional coupling had to be capacitive.

The signal generators were connected to the grid of the first limiter and the oscilloscope was connected to the second limiter grid leak. The setup was then adjusted to give a good trace on the screen. A 2.5 $\mu\mu$fd. ceramic capacitor was connected from the first limiter plate to the second limiter grid. The transformer response increased to over 300 kc. at the 3 db points. This showed that 2.5 $\mu\mu$fd. was too much coupling and, having no smaller capacitors, it was necessary to use a "gimmick." A short length of wire was soldered to the grid terminal of the transformer and the other end brought near to the plate terminal. The response of the transformer narrowed considerably. This showed that this capacity was in opposition to the magnetic coupling; the 2.5 $\mu\mu$fd. unit had completely overcome the magnetic coupling and had over-coupled the windings as well. To get the magnetic and capacitive coupling to aid, it was necessary to reverse the connections on one of the transformer windings. In the tuner being modified it was simplest to cross the connecting wires going to the plate and screen of the tube driving the primary of the transformer.

Once the windings were correctly polarized, the gimmick was reconnected to the grid terminal. Hooking the insulated end over the plate terminal gave sufficient capacity to provide the desired bandwidth. The transformer was now over-coupled, that is, the response curve was double-humped with a deep valley in the center. The valley was removed by loading both windings with 39,000-ohm resistors.

Working toward the front of the receiver, the same treatment was applied to each of transformers. As the first transformer was adjusted the influence of feedback from the output of the i.f. system became troublesome. The trouble was temporarily eliminated by connecting the scope to the grid resistor of the first limiter and removing the second limiter tube from the socket. This permitted modification of the first transformer but, of course, some other remedy was needed before the system as a whole could be aligned. In this case the trouble was corrected by connecting a choke (consisting of 15 turns of #20 insulated wire, wound on a ¼″ form) between the hot side of the heaters on the first limiter and the preceding tube and connecting a 5000 $\mu\mu$fd. disc ceramic capacitor from the hot heater terminal of the tube preceding the first limiter to ground. It may also be necessary to choke and bypass the plate supply leads and shield the bottom of the discriminator transformer and the discriminator socket.

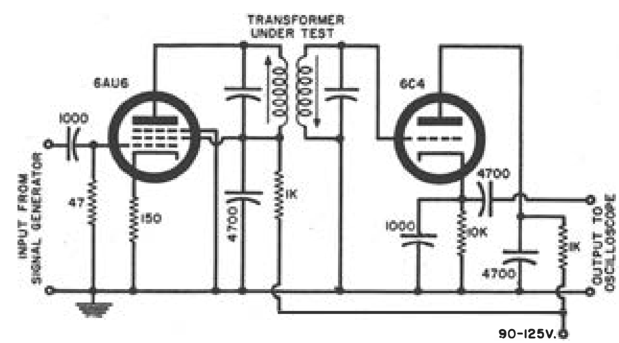
The presence of this undesirable feedback is manifested by a wide variation in the response curve when the strength of the input signal is varied. When the signal is strong, the gain of the limiters is decreased and the feedback causes no trouble. When the input signal level is reduced, the gain of the limiters increases and the feedback also increases. When it is of considerable magnitude, it has a great effect on the response curve of the amplifier. It ordinarily results in a peak in the curve at low signal levels with the peak disappearing as the level is increased. The presence of this feedback can be checked by shorting out the primary of the discriminator transformer and noting whether the response, up to either of the limiter grids, is affected. If the curve is appreciably changed when the primary of the transformer is shorted, you have work to do.

After the feedback was reduced to a satisfactory level, the i.f. system was aligned. Fig. 1E shows the i.f. response of the modified tuner. At the second limiter grid the bandwidth is 200 kc. with the ± 100 kc. points less than 2 db down. More important is the flat response across most of the 200 kc. bandpass. The discriminator curve, after modification of the rest of the i.f. transformers, showed that the discriminator transformer itself had sufficient bandwidth and thus needed no modification. Fig. 1F shows the discriminator curve of the modified tuner. The peaks are now separated 400 kc. and the line between them is straight. Listening tests showed that any station which gave limiting could be received satisfactorily.

The procedure for modifying a ratio-detector-type tuner with a limiter is the same as for the limiter-discriminator type. The modification of a ratio-detector type which has no limiter requires that some arrangement be made to check the response of the i.f. stages independent of the response of the ratio detector stage. In some cases it will be possible to get the signal for the scope from the screen of the driver tube. If, in the tuner you are checking, it is not possible to find any point which will provide a scope signal, it will be necessary to use a probe, similar to that shown in Fig. 4, to check out the i.f. system.

The changes described took only a few hours to make. This modest amount of work has paid big dividends in listening satisfaction. If you have or can borrow the necessary test equipment, you can make these same improvements in your tuner.

Fig. 2. The schematic diagram shown here is of a circuit that may be employed for the testing of individual i.f. transformers as is described in text.



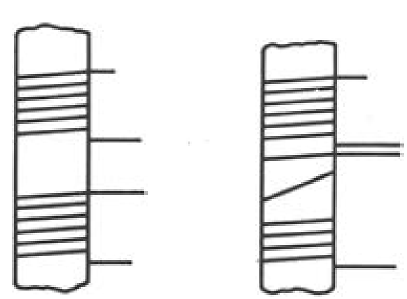


Fig. 3. I.f. windings before and after the coupling was increased. See text.

Fig. 4. Special oscilloscope probe for use where i.f. system has no limiters.

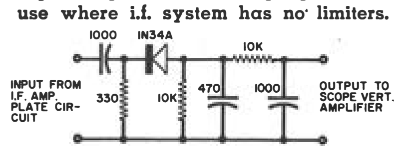
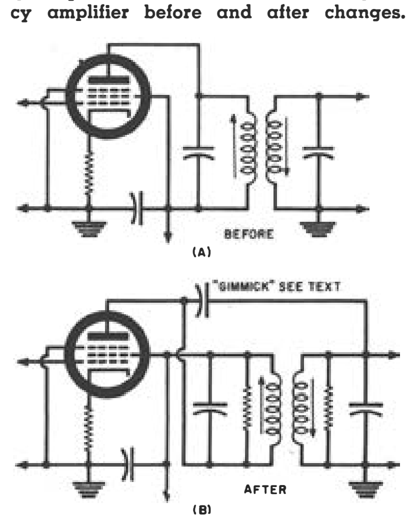


Fig. 5. Illustrations show a typical frequency-modulation intermediate frequency amplifier before and after changes.



Author Jones is shown with the stereo system described below. The two preamps are at the bottom of the tape deck carrying case, the two 10-watt amplifiers are on the shelf between the two speakers, and the common power supply is at the left.

Complete construction data on a semiconductor stereo system which provides 20 watts of hi-fi power output.

By **DWIGHT V. JONES**

All-Transistor Stereo Tape System

EDITOR'S NOTE: *For those of our readers who are interested in constructing the unit described below, we would like to point out that there are nine transistors used in each of the two channels, plus four silicon rectifiers in the power supply. The four power transistors are* Delco *types available directly from local* Delco *distributors at about $6.00 apiece. The other transistors and rectifiers are readily available and are less expensive. The total cost of semiconductors alone may run between $70.00 and $80.00 for this unit.*

In spite of the above, however, we feel that this article is important even to those who may not duplicate the circuit since it shows what can be done with transistors and transistor circuitry at this time.

TWENTY watts of electrical energy is available from this system for driving your living room speakers; also adequate tone controls to compensate for variations in components, program material, and the human ear at different listening levels. This system consists of a stereophonic tape deck, two tape preamplifiers, two 10-watt amplifiers, two 8- or 16-ohm speaker systems, and a common power supply, as indicated in the block diagram of Fig. 7.

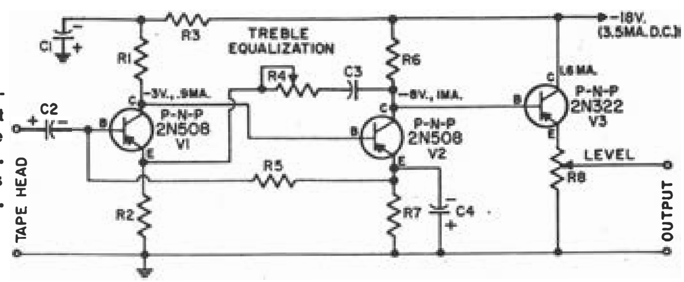
Transistors offer many advantages in high-fidelity circuits since there is no problem with microphonics or hum pickup from filaments as we have with tubes. Transistors are inherently low-impedance devices and thus offer better matching to magnetic pickups and loudspeakers, for more efficient power transfer.

Preamplifiers

The two preamps consist of two identical units with the circuitry of Fig. 1. Both preamplifiers use a common 18-volt battery supply. The circuit of Fig. 1 consists of three direct-coupled transistor stages. The first two stages have a feedback bias arrangement for current stabilization of the two stages. The 330,000-ohm resistor from the emitter of V_2 provides this d.c. current feedback to the base of V_1. The output stage is well stabilized with a 5000-ohm emitter resistance, R_8.

The negative feedback from the collector of V_2 to the emitter of V_1 is frequency selective to compensate for the standard NAB (formerly NARTB) recording characteristic. The preamplifier frequency response from a record-

Fig. 1. Complete schematic diagram and part listing of one of the tape preamplifiers employed. Note two identical units are required for system.



R_1—15,000 ohm, ½ w. res.
R_2—47 ohm, ½ w. res.
R_3—1500 ohm, ½ w. res.
R_4—25,000 ohm linear taper pot ("Treble Equalization")
R_5—330,000 ohm, ½ w. res.
R_6—10,000 ohm, ½ w. res.
R_7—3000 ohm, ½ w. res.
R_8—5000 ohm audio taper pot ("Level Control")
C_1, C_2—20 µfd., 20 v. elec. capacitor
C_3—.01 µfd. ceramic capacitor
C_4—100 µfd., 6 v. elec. capacitor
V_1, V_2—"p-n-p" transistor (G-E 2N508)
V_3—"p-n-p" transistor (G-E 2N322)
NOTE: This listing is of parts required for ONE of the preamps. For the complete stereo system as described by the author, TWO such units must be built and the above parts duplicated.

Fig. 2. Response curves for the tape preamplifier unit.

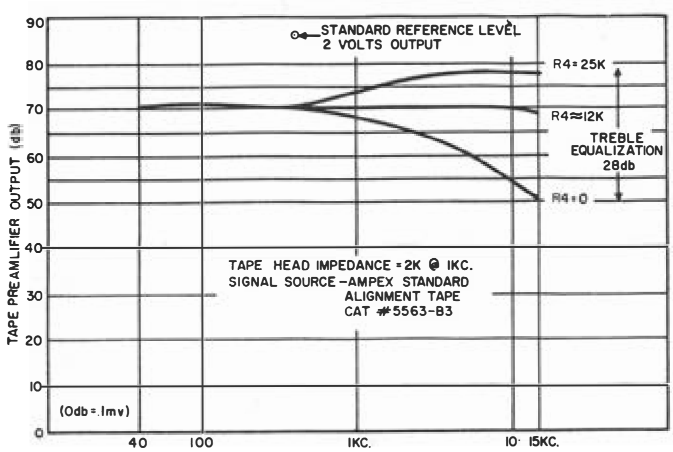
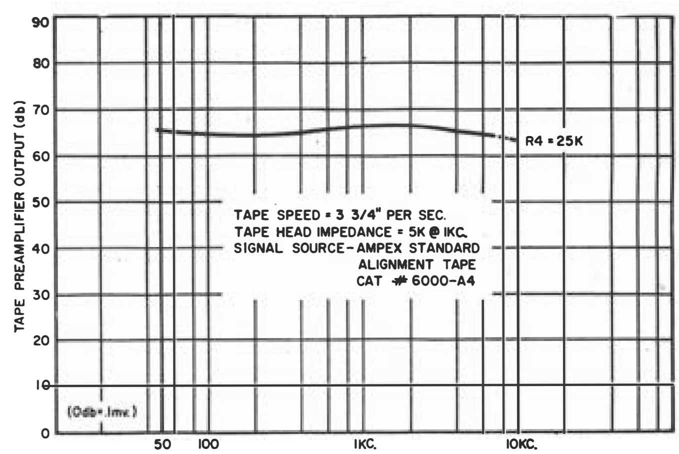


Fig. 3. Response of preamp altered for 3.75 ips tape speed.



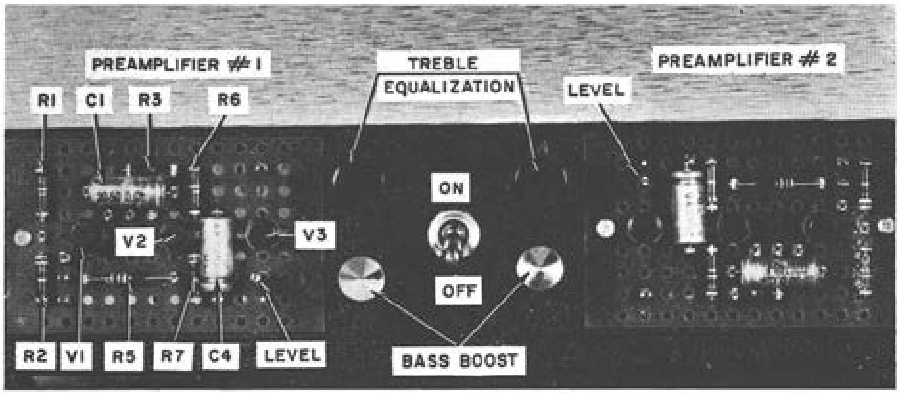
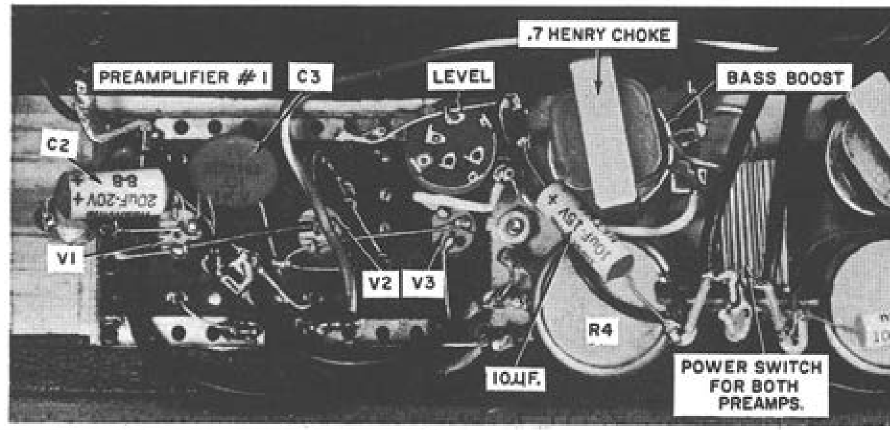


Fig. 4. Both tape preamps along with tone control circuits are shown here.

Fig. 5. Close-up view of rear of one preamp and the tone-control circuits.

ed tape at 7.5 ips is shown in Fig. 2. The flat response from a standard recorded tape occurs with the treble control, R_4, at mid-position or 12,000 ohms. There is 7 to 8 db of treble boost with the control at 25,000 ohms maximum position and approximately 20 db of treble cut with R_4 equal to zero.

The preamp output is approximately 2 volts with the input being the maximum 400-cycle recorded level for 2 percent distortion on the tape (Standard Reference Level). The total harmonic distortion of the preamp at this level is under .2 per-cent. With a preamp output of 4 volts at 400 cycles, the total harmonic distortion is still less



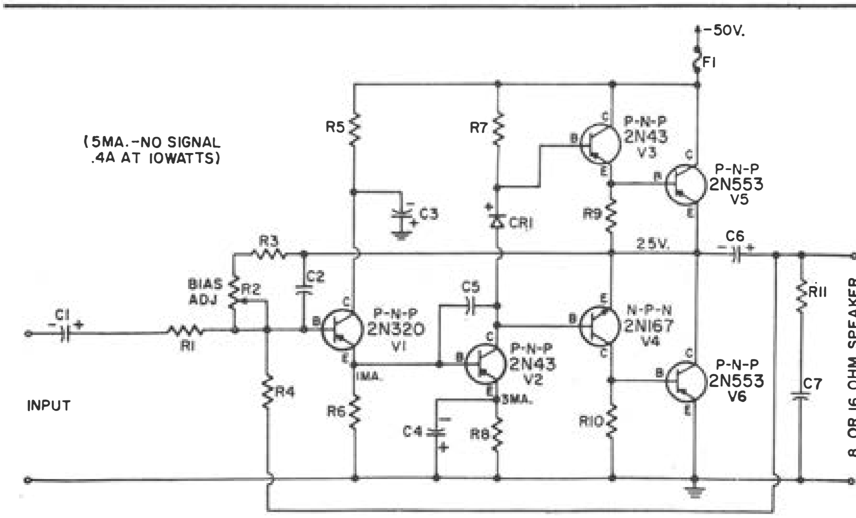
than 1 per-cent as actually measured.

This preamp will accommodate a variety of tape head impedances since it gives an equalized output for a 2000-ohm head at 1000 cps and also a 6000-ohm head. The input impedance of the preamp increases with frequency because of the frequency selective negative feedback to the emitter of V_1. The impedance of the tape head also increases with frequency but is below that of the preamp. The input impedance of the preamplifier is approximately 70,000 ohms at 1000 cps.

The 2N508 was used in this preamp because, basically, it is a high-gain low-noise transistor. This transistor operates in a circuit designed to achieve a good signal-to-noise ratio (S/N). The S/N of this preamp is approximately 60 db. The noise level will vary with respect to head structure, shielding, and physical layout of the tape deck, amplifiers, etc.

The emitter-follower stage gives a low-impedance output for a cable run to the power amplifier and acts as a buffer so that any preamp loading will not affect the equalization characteristic. A linear taper was used for R_4, the treble control, and an audio taper for R_8, the level control.

The preamplifier of Fig. 1 may be altered to compensate for tapes re-



R_1—3000 ohm, ½ w. res.
R_2—100,000 ohm pot ("Bias Adjust")
R_3—150,000 ohm, ½ w. res.
R_4—24,000 ohm, ½ w. res.
R_5—39,000 ohm, ½ w. res.
R_6—1500 ohm, ½ w. res.
R_7—8200 ohm, ½ w. res.
R_8—470 ohm, ½ w. res.
R_9, R_{10}—1000 ohm, ½ w. res.
R_{11}—22 ohm, ½ w. res.
C_1—20 μfd., 20 v. elec. capacitor
C_2—100 μμfd. ceramic capacitor
C_3—25 μfd., 50 v. elec. capacitor
C_4—100 μfd., 6 v. elec. capacitor
C_5—.001 μfd. capacitor
C_6—1000 μfd., 50 v. elec. capacitor
C_7—.2 μfd. paper capacitor
CR_1—1N91 germanium diode
F_1—½ amp. fuse
V_1—"p-n-p" transistor (G-E 2N320)
V_2, V_3—"p-n-p" transistor (G-E 2N43)
V_4—"n-p-n" transistor (G-E 2N167)
V_5, V_6—"p-n-p" transistor (Delco 2N553)
NOTE: This listing is of parts required for ONE of the amplifiers. For the complete stereo system as described by the author, TWO such units must be built and the above parts duplicated.

Fig. 6. Complete schematic of one of the two identical 10-watt power amplifiers.

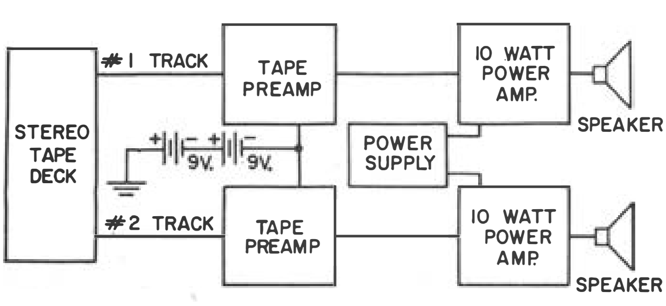
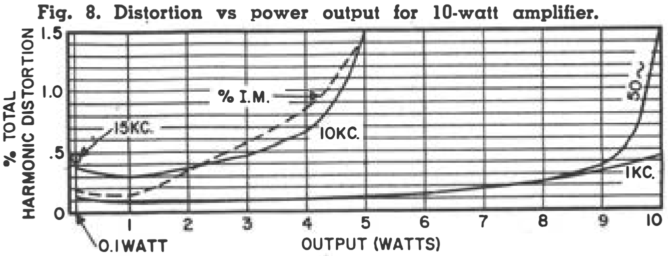


Fig. 7. Block diagram of the complete stereo tape system.

Fig. 8. Distortion vs power output for 10-watt amplifier.



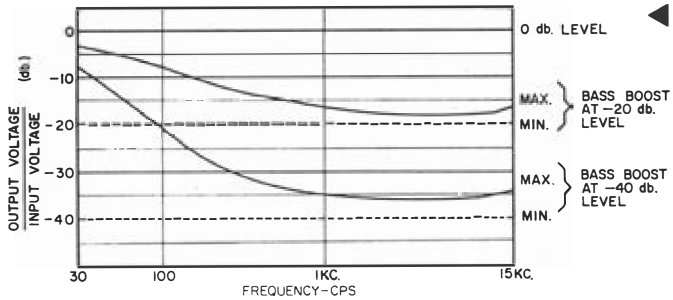
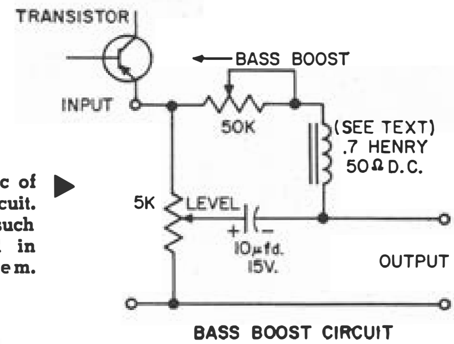


Fig. 9. Curves at the left show the frequency response of the bass-boost circuit employed at −20 and −40 db.

Fig. 10. Schematic of the bass-boost circuit. Note that two such circuits are used in the stereo system.



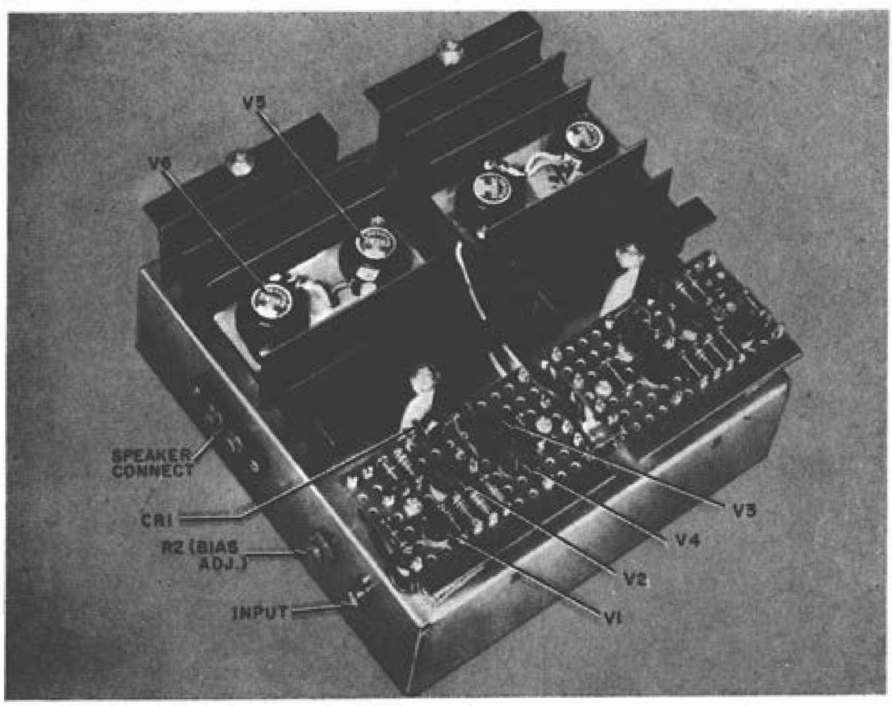


Fig. 11. Top view of the two 10-watt transistorized power amplifiers.

corded at 3.75 ips by setting R_4 at 25,000 ohms and making the feedback capacitor (C_3) .02 μfd. In addition, the 47-ohm resistor (R_2) from the emitter of V_1 to ground was shunted with .5 μfd. to attain the response shown in Fig. 3. The value needed for this shunt capacitor will depend somewhat on the high-frequency response of the tape head being used, since this capacitor contributes to increased circuit gain above 3000 cps.

Bass-Boost Circuit

A bass-boost circuit using three passive components in conjunction with the level control is included in the design. This gives the operator independent control of the level or amount of bass boost desired or the level control can be used as a loudness control. This circuit has the advantage of simplicity, economy, and has its best application in transistor circuits since it requires a very low driving impedance.

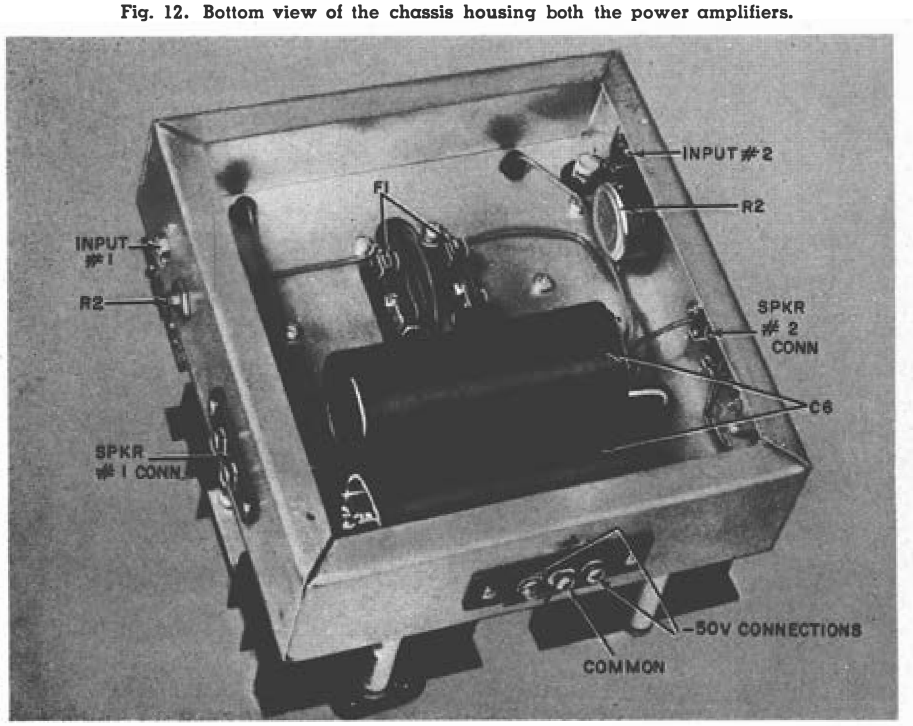
It is usually desirable to have some method of boosting the level of the lower portion of the audio spectrum as the over-all sound level is decreased. This is to compensate for the non-linear response of the human ear as represented by the now-familiar Fletcher-Munson curves. The ear requires a higher level for the low-frequency sound to be audible as the frequency is decreased and also as the over-all spectrum level is decreased.

The usual circuits that are employed to accomplish this attenuate the over-all audio spectrum independent of the level control and then with a variable network the low-frequency attenuation is decreased—giving bass boost. This means that circuit gain has been sacrificed to obtain bass boosting. Frequency sensitive feedback around one or more stages is another method used to accomplish bass boost.

The new simplified bass-boost circuit shown in Fig. 10 gives the desired result without sacrificing circuit gain or adding gain stages.

This circuit is more easily adapted to transistor circuitry since it requires a very low drive impedance for optimum performance. It is quite practical to attain this low impedance (less than 50 ohms) from a transistor in the emit-

Fig. 12. Bottom view of the chassis housing both the power amplifiers.



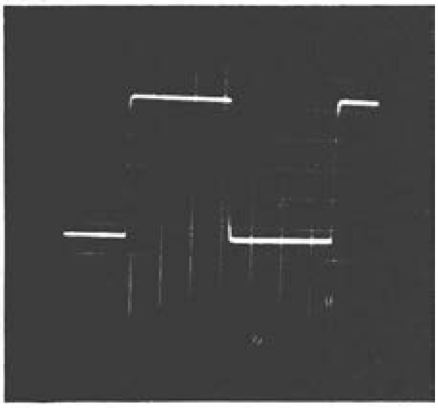


Fig. 13. One-kc. square-wave response.

ter-follower connection, as shown in Fig. 10.

Fig. 9 shows the frequency characteristics of this circuit. With the level control set for zero attenuation at the output there is no bass boost available, but as the output level is attenuated, the available bass boost increases.

Fig. 9 shows the frequency response (lower dashed curve) when the output is attenuated 40 db and the bass-boost control is set for minimum (50,000 ohms). The solid curve immediately above represents the frequency response when the bass-boost control is set at maximum (zero ohms). Thus a frequency of 30 cps can have anything from zero to 27 db of boost, with respect to 1000 cps, depending on the adjustment of the bass-boost control.

All components used in the construction are standard with the exception of the inductance, which weighs about 1 ounce and has over-all dimensions of about 1" x ¾" x ¾". The coil is wound on standard nickel-steel laminations that are used for transistor transformers. The author obtained the 0.7-henry inductance by using the green and yellow leads on the secondary of *Argonne* transistor transformer No. AR-128.

This circuit about fulfills the requirements set forth in the article "Is a Loudness Control Necessary" by Burt Hines published in the July 1958 issue of Radio & TV News. This article indicated a separate loudness control would not be needed if a bass control could provide "something like 25 to 30 db at 40 cycles." Fig. 9 shows that approximately 25 db of boost can be obtained with this circuit. The Fletcher-Munson contours of equal loudness level show most of the contour changes involve a boost of the bass frequencies at the lower levels of intensity. Therefore, this circuit combination seems to fulfill the requirements of level control, bass boost, and loudness control.

The circuit can be added to the emitter-follower and level-control output of Fig. 1 since it has the same driving impedance, as shown in Fig. 10.

With this addition, the preamp of Fig. 1 now has the necessary treble and bass control to compensate for listening levels, deficiencies in the program material, or deficiencies in pickup, speakers, etc.

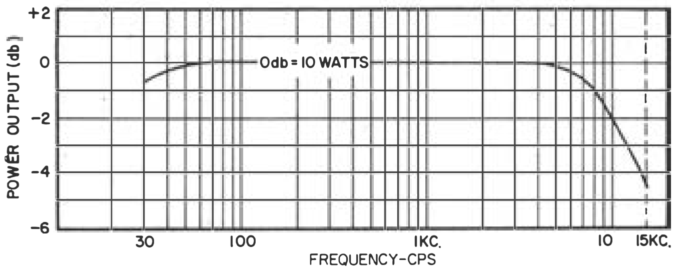
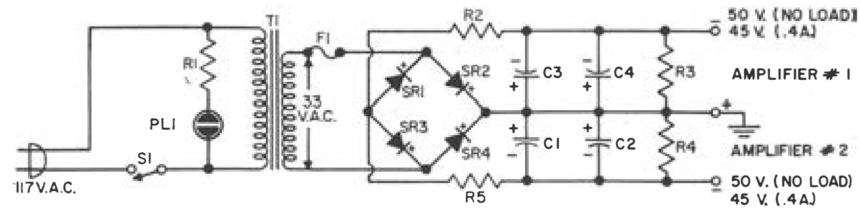


Fig. 14. The curve at the right shows the maximum power response of one of the power amplifiers at the level at which clipping begins.



R_1—180,000 ohm, ½ w. res.
R_2, R_5—2 ohm, 5 w. res.
R_3, R_4—47,000 ohm, ½ w. res.
C_1, C_2, C_3, C_4—1500 µfd., 50 v. elec. capacitor
SR_1, SR_2, SR_3, SR_4—Silicon rectifier (1N1115)
T_1—Rectifier power trans. 117 v. pri.; 33 v. @ 3 amp sec. (Triad F-61U or equiv.)
F_1—2 amp fuse S_1—S.p.s.t. switch
PL_1—1/25th watt neon bulb (NE-51)
NOTE: This supply will power both amplifiers.

Fig. 15. Complete schematic diagram of the single power supply unit required.

tening levels, deficiencies in the program material, or deficiencies in pickup, speakers, etc.

The construction and component layout of the two preamplifiers with this bass-boost circuit is shown in Fig. 4. Fig. 5 shows the construction on the back side of one of the preamps.

Power Amplifiers

A great deal of effort has gone into the development of transformerless push-pull amplifiers using vacuum tubes. Practical circuits, however, use many tubes in parallel to provide the high currents necessary for direct driving of low-impedance speakers.

The advent of power transistors has given new impetus to the development of transformerless circuits since transistors are basically low-voltage, high-current devices. The emitter-follower stage, in particular, offers the most interesting possibilities since it has low inherent distortion and low output impedance.

The two 10-watt power amplifiers used in this system consist of two identical amplifiers with circuitry as shown in Fig. 6. This is a direct-coupled amplifier with excellent low-frequency response and also has the advantage of a feedback arrangement for current stabilization of all stages. The feedback system also stabilizes the voltage division across the power output transistors, V_5 and V_6, which operate in a class B push-pull arrangement. V_3 and V_4 also operate class B in the familiar Darlington connection to increase the current gain. Using an *n-p-n* for V_4 gives the required phase inversion for driving V_6 and also has the advantage of push-pull emitter-follower operation. V_5 and V_6 have a small forward bias to minimize crossover distortion. This bias is set by the voltage drop across the 1000-ohm resistors (R_9 and R_{10}) that shunt the input to V_5 and V_6. V_3 and V_4 are biased for the same reason with the voltage drop across the 1N91. A 68-ohm resistor would serve the same function as the 1N91 except there would be no temperature compensation. Thermistors have also been used to compensate for the temperature variation of the emitter-base resistance, but they do not track this variation as well as a germanium junction diode which has temperature characteristics similar to the transistor.

V_2 is a class A driver requiring a very low-impedance drive which is accomplished by an emitter follower, V_1. V_1 needs a current source for low distortion thus R_1 was set at the maximum value which would allow the amplifier to be driven to maximum output with the Standard Reference Level into the preamp.

The bias adjust, R_2, is set for one-half the supply voltage across V_6 and can be trimmed for symmetrical clipping at maximum power output. The .001-µfd. feedback capacitor (C_5) from collector to base of V_2 aids in stabilizing this circuit by reducing the phase shift and high-frequency gain of this stage. The 100-µµfd. capacitor (C_2) shunting the bias network further aids the stabilization with high-frequency negative feedback from output to input. This circuit has approximately 15 db of over-all feedback with the 24,000-ohm resistor (R_4) from load to input. The speaker system is shunted by 22 ohms (R_{11}) in series with .2 µfd. (C_7) to prevent the continued rise of speaker impedance and its accompanying phase shift beyond the audio spectrum.

The over-all result from using direct-coupling, no transformers, and ample degeneration, is an amplifier with output impedance of ½ ohm for good speaker damping and very low total harmonic distortion, as shown in

Fig. 16. Top view of the common power supply chassis.

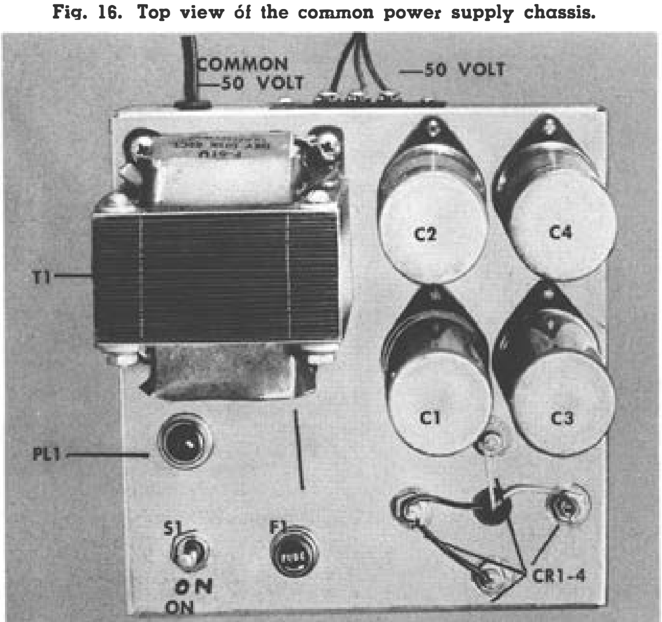


Fig. 17. Bottom view of the common power supply unit.

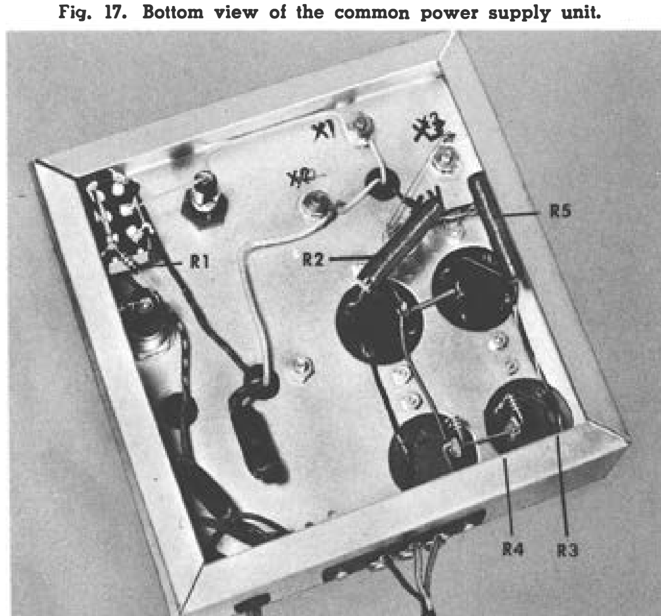


Fig. 8. The frequency response at average listening levels is flat over the audio spectrum and the maximum power response is shown in Fig. 14. The amplifier square-wave response is shown in Fig. 13. All of the data was obtained using an amplifier load of 16 ohms.

Figs. 11 and 12 show the construction and layout of both 10-watt amplifiers on a single aluminum chassis. V_5 and V_6 of Fig. 6 are mounted on a common heat radiator which is insulated from the chassis. See Fig. 11. One of these transistors must be insulated from the common heat radiator. The author placed a thin sheet of mica between transistor V_6 of both amplifiers and the *Delco* automobile radio heat sink. These aluminum heat sinks could have been sawed in half for direct mounting of each of the four power transistors. Another method would be to mount each power transistor directly on a 3" x 3" x 3/32" aluminum plate, with each of the four plates insulated from each other and also insulated from the chassis.

The two 10-watt amplifiers use the common power supply of Fig. 15 which provides excellent isolation for each amplifier. Figs. 16 and 17 show the layout and construction of the power supply used by the author. Mounting hardware and mica washers are furnished with the 1N1115 silicon rectifiers. The aluminum power supply chassis (Fig. 16) is the heat radiator for the bridge rectifier system that is used to deliver the required d.c.

Although the construction described may seem fairly complicated and be relatively expensive, the results obtained have fully justified the author's outlay of time and effort.

"STEREO" DEFINED

APPROVAL of a concise and complete definition of the word "stereophonic" has been voted by the Board of Directors of the Magnetic Recording Industry Association.

The definition was formulated by the Standards Committee of MRIA, headed by C. J. LeBel, and after approval by the association's board became standard for the magnetic recording industry. The definition will be forwarded to the American Standards Association for its consideration. The definition is as follows:

"Stereophonic, stereo (binaural, deprecated): A technique of transmitting sound which employs two or more complete transmission channels for the purpose of creating in the listening environment the sense of auditory perspective inherent in the source environment. Each channel must include a separate microphone, amplifier, and loudspeaker, and may have one channel of a multichannel recorder and reproducer interposed as a time-storage device."

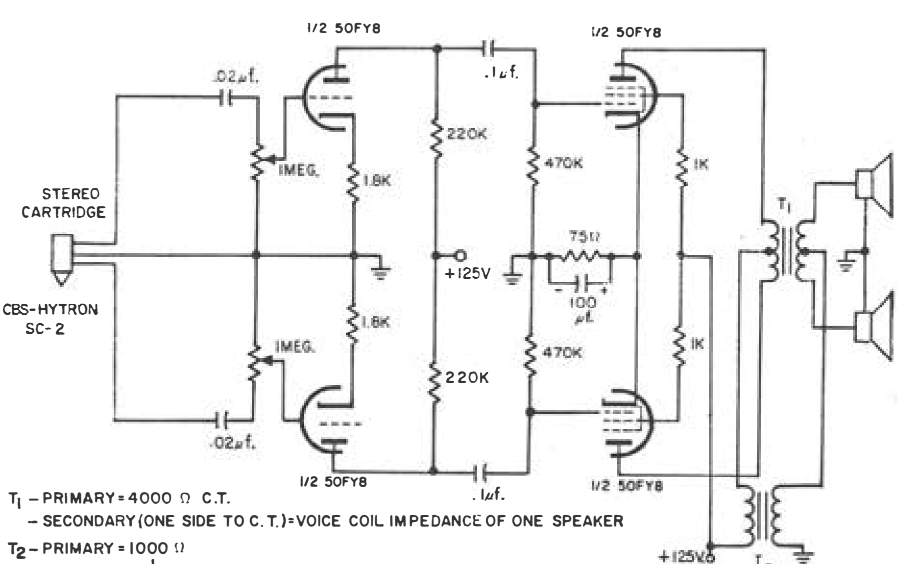
2-Tube 7-Watt Stereo Amplifier

A NEW miniature triode-pentode that makes possible very compact audio amplifiers, has just been introduced by CBS Electronics. The 50FY8 combines the voltage and power amplifiers in a single envelope. In addition to the 50-volt heater type, 25-, 12.6- and 6.3-volt heater types are also available as the 25FY8, 12FY8, and 6FY8, respectively. The tube features low plate and screen voltage requirements and provides 2.7 watts output in single-ended class A.

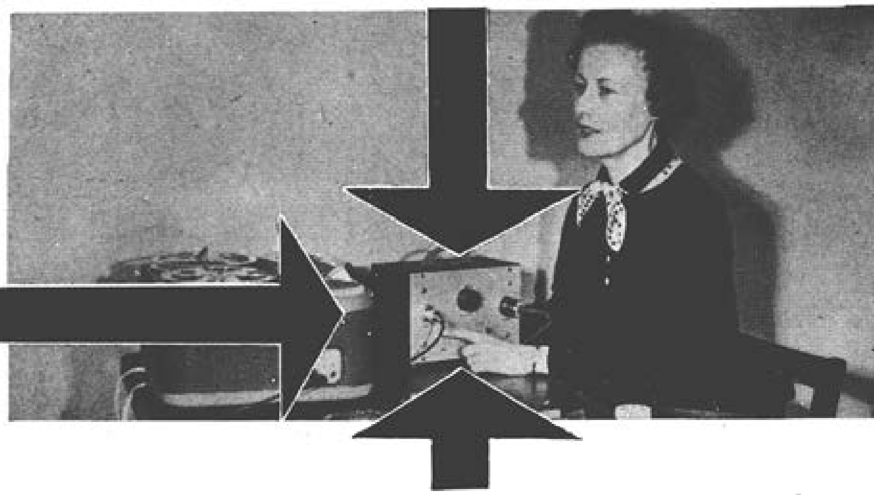
A pair of 50FY8's are the only tubes required in a stereo amplifier utilizing the CBS modified simplex circuit. Such an amplifier can provide up to 3.5 watts per channel. Complete construction details of an amplifier using two of these tubes, along with tone-control circuits and power supply, will appear in our September issue under the title "Compact Two-Tube Stereo Amplifier."

The technical bulletin E-334 describing the new audio tube is available from CBS Electronics Advertising Service, Parker St., Newburyport, Mass.

Schematic diagram of the two-tube, seven-watt stereo amplifier using 50FY8's.



By **HAROLD REED**



Listening to tape playback and editing with aid of the switch, which is being operated by 20 cps pulses on the tape itself.

20 cps Tape Recorder Switch

Use low-frequency signal recorded on tape to operate power circuits through this automatic switching unit.

ALTHOUGH designed primarily for operating a relay by means of a low-frequency signal recorded on magnetic tape, this device may be used to operate a relay for control applications when any 20-cycle, or lower, signal is applied to its input. It may be used to stop the recorder actually being used to play back the tape, to start or stop another recorder, to control a motion picture projector, in fact to control any electrical device or electric lamps in demonstration applications.

The control signal can be obtained from any audio oscillator capable of supplying the low-frequency signal. This signal may be used to actuate the device directly or it can be recorded on magnetic tape and the unit operated during tape playback once, or as many times as desired and for any time duration.

The filter has a sharp cut-off characteristic above 10 cycles, therefore, a control signal between approximately 10 and 20 cycles will operate the switch, but voice, music, and sine-wave signals of higher frequencies will not trigger the unit. This makes it possible to feed the output of a tape recorder to the unit and only the low-frequency control voltage will cause the switch to operate and only at each point that this control signal is recorded on the tape.

A self-contained germanium diode power supply is included but the circuit may be operated from a 22½-volt battery if desired.

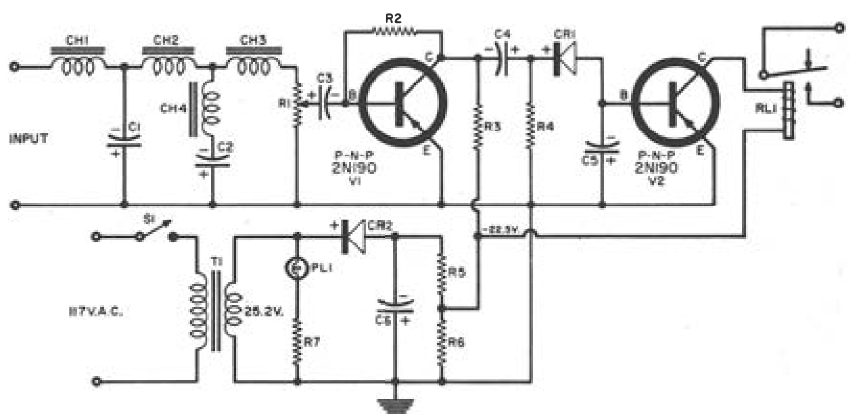
Circuit Description

The device employs a low-pass filter in the input circuit, as shown in Fig. 1. This filter was designed to operate at the lower end of the audio band, that is, 20 cycles or below. Using a low frequency makes it possible to record this signal on a magnetic tape along with voice or music and the low-frequency control signal will trigger the relay but will be inaudible in the loudspeaker. Mathematical equations used in designing the filter are discussed later. Unless the constructor is especially interested, or wishes to design the filter for another frequency or impedance, he may ignore such design data.

A common-emitter transistor audio amplifier follows the filter section. This amplifier works into a germanium diode rectifier. Direct-current output from the diode is applied to a transistor d.c. amplifier with the sensitive relay connected in its collector circuit.

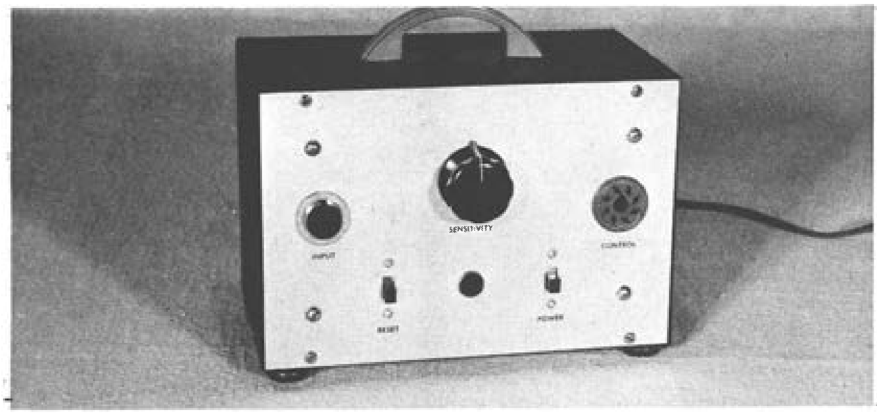
Voltage divider R_5 and R_6 allows for adjustment of the power-supply output to −22.5 volts. These resistors also provide a bleeder load for the power supply. Since the circuit was designed for this supply voltage, it is possible to build the unit with either or both an a.c. and a 22.5-volt battery supply and switch in either as required. As there is no 6.3-volt winding available, the power pilot lamp is connected across the 25-volt winding in series with dropping resistor R_7. The pilot lamp and resistor may be omitted.

Although two relays are employed in the unit shown, the sensitive relay, RL_1, can be used alone to operate the recorder motor or other device. It has a contact rating of 2 amperes. The second relay, RL_2 of Fig. 2, is an ordinary 117 volt a.c. type. It is wired to provide a self locking-in action, elimi-



R_1—1000 ohm pot
R_2—150,000 ohm, ½ w. res.
R_3—10,000 ohm, ½ w. res.
R_4—470 ohm, ½ w. res.
R_5—100 ohm, ½ w. res.
R_6—600 ohm, ½ w. res.
R_7—150 ohm, 5 w. res. (optional, see text)
C_1, C_5—25 µfd., 25 v. elec. capacitor
C_2—16 µfd., 12 v. elec. capacitor
C_3, C_4—10 µfd., 25 v. elec. capacitor
C_6—16 µfd., 50 v. elec. capacitor
CH_1—4.5 hy. choke
CH_2—8 hy. choke
CH_3, CH_4—2.5 hy. choke
PL_1—#47 pilot light (optional, see text)
S_1—S.p.s.t. switch
T_1—Fil. trans. 25.2 v. @ 1 amp. (Stancor P6469 or equiv.)
RL_1—8000-ohm relay (Sigma 4F)
CR_1—1N34 crystal diode (Raytheon)
CR_2—1N38A crystal diode (Sylvania)
V_1, V_2—"p-n-p" transistor (G-E 2N190)

Fig. 1. Schematic diagram and parts listing of the transistorized 20 cps switch.



Front-panel view of the completed unit shown in its metal case.

nating the need for a more expensive latching-type relay. Release of the relay to the normal, or open, condition is easily accomplished with reset switch, S_2. Thus, relays RL_1 and RL_2 may be used to start or stop any device and RL_2 may be selected to handle greater loads.

The value of the sensitivity control at the input of the amplifier was selected so that it would be at about midway position when a control signal of 1 volt was applied to the filter input. A 1000-ohm potentiometer proved to be about right for this purpose.

Impedance mismatch at the filter output varies with movement of the sensitivity control. For this application, however, we are not concerned with precise impedance matching as long as we obtain a low-frequency signal of sufficient amplitude at the base of V_1. A more expensive constant-impedance attenuator could be used here but was not considered necessary.

Construction

Although a standard 5" x 6" x 9" metal box was used to house the device, it could be built into a much smaller container. The size shown, however, will provide room for a battery and change-over switch, if both battery and a.c. operation are desired, with space to spare.

The input connector, control connector, power and reset switches, pilot lamp, and sensitivity control are mounted on the front panel. The low-pass filter section, power supply, relays, transistor sockets, and other component parts are assembled on a 7½" x 4½" sub-panel which is attached to the main panel by means of screws and four spacer posts. These posts may be of any suitable material. The author used threaded ceramic insulators.

Stand-off terminal strips were mounted on the inside of the sub-panel under the machine screws holding the chokes, etc., as required. Capacitors, resistors, and the crystal diodes were soldered to these terminals between the sub- and front panel. Miniature transistor circuitry capacitors were used. All wiring on the sub-panel must be completed first and then the interconnecting leads between it and the front panel soldered in place just before the sub-panel is installed. The constructor need not follow this type of layout, however, since parts placement is non-critical.

Both panels are aluminum sheets and the front panel was etched and provided with identifying decals under the controls and connectors. The box was fitted with a plastic handle and four rubber feet. Most of the items used in the unit shown in the photos were found in the junk box. The parts list gives commercial components that will provide equivalent results.

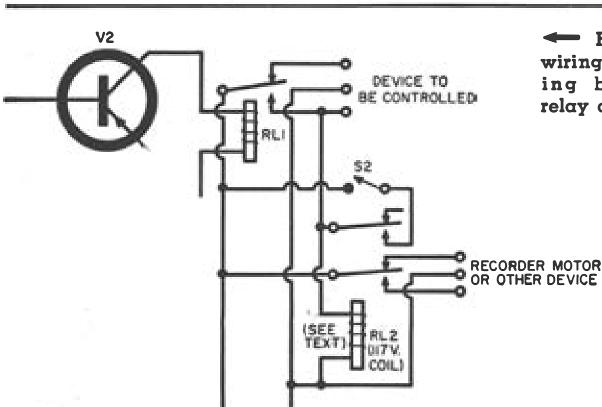
Incidentally, after cutting transistor leads to a short length, it is difficult to plug them into the socket. The author found that if after the leads are cut, they are filed lightly with a fine file to taper the lead ends, they can be inserted into the sockets quite easily.

As mentioned previously, the following design data is given for the benefit of the reader interested in constructing filter networks with characteristics other than those of the author's "model."

The filter consists of a constant-k "T" section and an m-derived "T" section. The equations for the constant-k section are: $L_k = R/\pi f_c$ and $C_k = 1/\pi f_c R$. The circuit diagram is shown in Fig. 3A. Equations for the m-derived sections are: $L_1 = mL_k$, $L_2 = (1-m^2/4m)\ L_k$, and $C_1 = mC_k$. Refer to the diagram of Fig. 3B. In the foregoing equations, L is in henrys, C is in farads, R equals the characteristic impedance, π is 3.14, f_c is the cut-off frequency, m is 0.6. The m value of 0.6 is commonly used in practical filter design. Its value is found by $m = \sqrt{1-(f_c/f_\infty)^2}$ where f_∞ is a frequency of high attenuation. The inductance L_k in the output leg of the constant-k section and the input inductance L_1 of the m-derived section are combined, that is, the total value of these two inductances is furnished by a single coil. This is shown in Fig. 5 which is the schematic diagram of the complete filter as designed from the preceding equations.

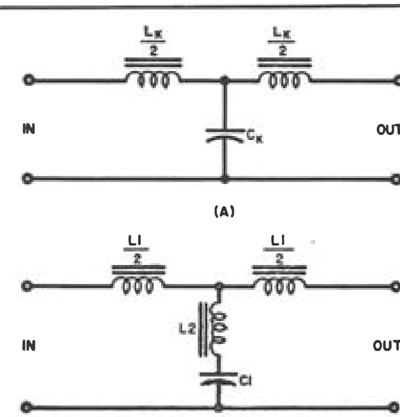
For extremely efficient filter operation and in critical applications, top quality parts are required. The resistance of the coils should be low and the "Q" of the coils and capacitors high. However, for this application and at the low frequency involved, the author found ordinary iron-core choke coils and electrolytic capacitors to be satisfactory. Commercially available choke coils and capacitors having inductance and capacitance values as close as possible to the values derived from the equations were chosen. These are given in Fig. 1 and in the parts list. Note how close the values of the commercially available items come to the filter design values. A response curve of the filter, using the specified parts, is shown in Fig. 6. Measured values are given in Table 1.

Two methods for checking the filter response are presented in Fig. 4. When a 600-ohm, high-output oscillator is available, the filter can be checked as shown in Fig. 4A. The oscillator output voltage must also be constant over the frequency range. If the oscillator out-



← Fig. 2. Schematic wiring diagram for using both sensitive relay and lock-in relay.

Fig. 3. (A) Configuration of constant-k "T" section and (B) m-derived "T" section. →



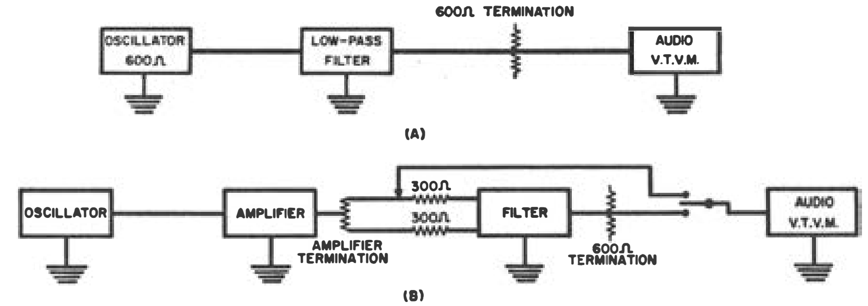
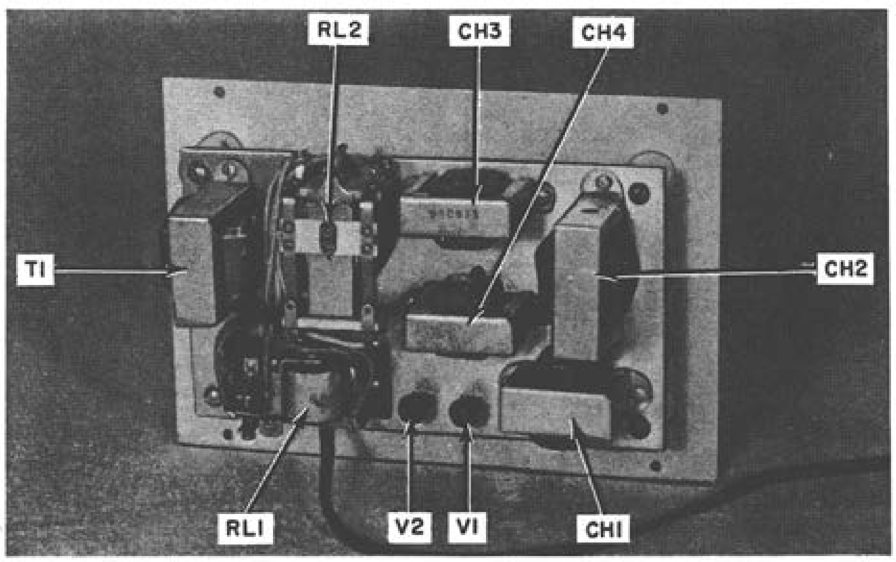


Fig. 4. Checking filter response with (A) high and (B) low output oscillator.



Rear view showing sub-panel attached to main panel of the unit.

put is too low for this arrangement, the test setup of Fig. 4B may be used. The input voltage to the filter is held constant by switching in an audio vacuum-tube voltmeter for each test frequency. If the amplifier output is known to be completely flat over the frequency range, the voltmeter switch is not required. If the amplifier has an output impedance of 600-ohms it may be worked directly into the filter without the series resistors shown in Fig. 4B.

Voltage and Current Data

The following test data was taken when using an audio control signal of 20 cycles. With the snsitivity signal of wide open, that is, maximum sensitivity, the relay closed with an audio input signal of 0.17 volt at the filter input and opened when this signal voltage was reduced to 0.15 volt. With the sensitivity control at about midway position, the relay closed at 1 volt of audio signal to the filter input and opened when the signal was reduced to 0.7 volt.

V_1 collector voltage measured −5.6 volts and collector current was 1.7 ma. The V_2 collector voltage read −21 volts with no audio signal and −15 volts with the relay closed (1 ma. current flow). V_2 collector current with no audio signal was 0.2 ma., with 1 volt audio signal (sensitivity control midway) it was 1 ma. With the sensitivity control at maximum and an audio input signal of 0.5 volt, the V_2 collector current ran 2 ma.

Operating Data

To use this device with a tape recorder, the 20-cycle control signal is recorded on the tape at any spot at which it is desired to have the control device operate. This signal may be recorded separately or simultaneously with other recorded material and it may be recorded on the tape at as many points as desired. The control signal may be obtained from any oscillator and recorded on the tape through one of the regular recorder input channels.

The control unit is then connected to the output of the tape recorder. When the tape is played back, relay RL_1 will close at any time and for any time duration that the 20-cycle control signal is reproduced from the tape. This may be a single short pulse to momentarily operate the relay or it may be a prolonged signal to hold the relay closed for a definite period of time. When operating in this manner, switch S_2 may be in the "off," or open, position, so that RL_2 will open or close in consonance with RL_1. Thus, any device may be started or stopped momentarily or for prolonged periods, depending on duration of the 20-cycle control signal recorded on the tape.

Another mode of operation is to have relay RL_2 lock-in upon operation of RL_1. For this action, switch S_2 must be "on," or closed. Now a 20-cycle pulse applied to the input will actuate both relays as before, but RL_2 will latch in and remain closed even after the control signal has been removed. In this way, any controlled device may be started or stopped and will remain in this condition until relay RL_2 is reset by opening switch S_2.

If the mechanism of the recorder driving the tape that supplies the control signal is powered through relay RL_2, it too may be stopped automatically. This is a convenience in editing tapes. As an example, during the recording process a 20-cycle pulse may be superimposed on the other recorded material at any point where it is desired to make changes or to splice in other tape material. Then when the tape is played back for editing it will stop at each spot containing the low-frequency control signal and remain stopped until switch S_2 is reset.

Only several applications for the device have been suggested although it can be employed in numerous ways, not only by the audio experimenter and hobbyist, but commercially as well.

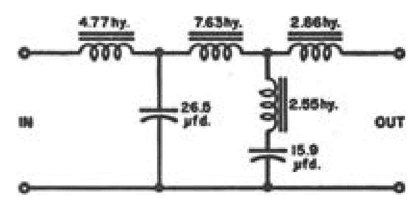


Fig. 5. Complete filter designed from text.

Fig. 6. Response of the filter network.

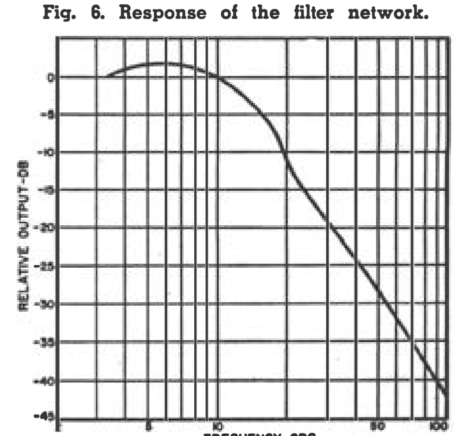
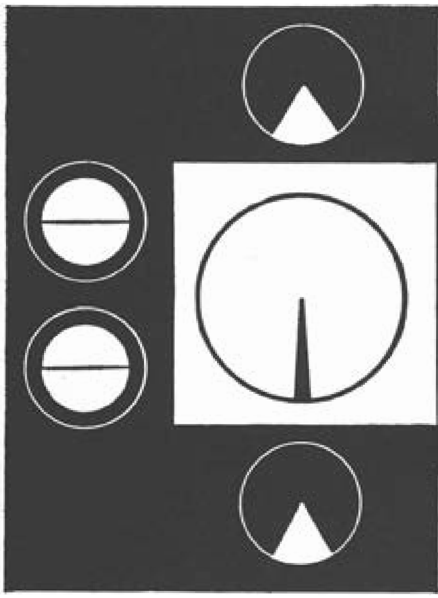


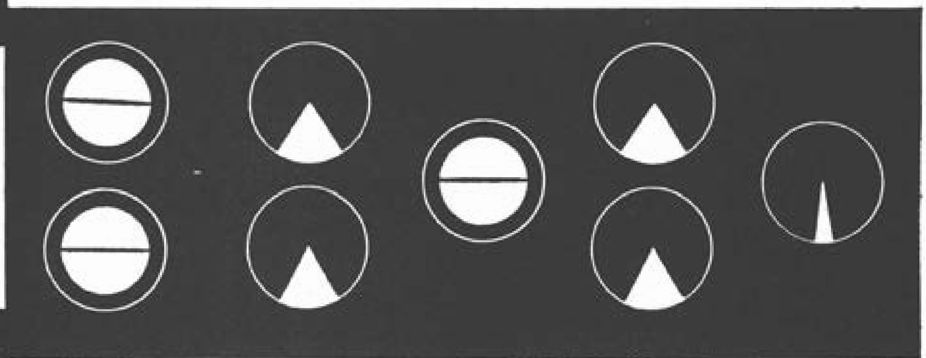
Table 1. Listing of relative outputs.

FREQUENCY (cps)	RELATIVE OUTPUT (db)
100	−42
90	−40
80	−38
70	−35
60	−32
50	−29
40	−24
30	−19
20	−11
10	0

Electronic Level Indicators For Tape Recorders

By **HERMAN BURSTEIN**





Helpful information on operation, circuits, and use of electron-ray tubes and neon lamps in home-type recorders.

A TAPE recorder should include an indicator of some sort to inform the user whether he is recording at a proper level or not, since excessive distortion will result from too high a level and an inferior signal-to-noise ratio from too low a level. Although a number of tape machines, particularly professional and semi-professional ones, employ vu and similar meters, electronic indicators are generally used in home-type recorders. This includes a number of tape machines suitable for high-fidelity applications.

Electronic indicators are of two kinds, the electron-ray ("magic eye") tube and the neon glow lamp. An understanding of how they operate, of their circuitry, of their advantages and disadvantages relative to meter indicators, and of the various problems associated with their use should be of value to the technician called upon to repair, adjust, or modify a home tape recorder. It should also be helpful to the audiophile desiring best results from his tape recorder, especially if he is inclined to tinker with the insides of his audio equipment from time to time. Unless the technician or audiophile understands electronic indicator circuitry, it is quite possible that in making a repair, adjustment, or change he may defeat the true purpose of the indicator.

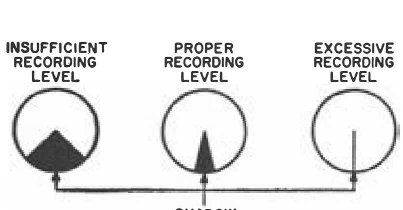
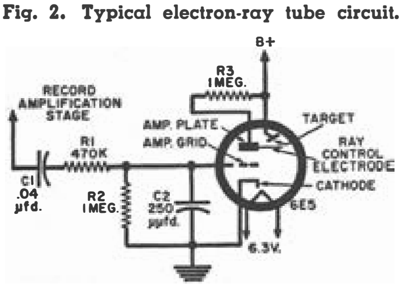


Fig. 1. Action of an electron-ray tube.

Fig. 2. Typical electron-ray tube circuit.



The Electron-Ray Tube

Action of the electron-ray tube—the 6E5 is probably the most popular type—is illustrated in Fig. 1. The audio signal fed to the grid of the tube causes the shadow of the eye to be just barely open when maximum permissible recording level is reached. Excessive recording level causes the eye to close or overlap, while insufficient level causes it to remain substantially or completely open.

When recording, the user tries to set gain to nearly close the eye on audio peaks. If the eye does close occasionally on a transient, there is probably no harm done but if the eye is closed much of the time, tape distortion will be manifest in playback.

The Neon Lamp

The less expensive recorders tend to employ a neon glow lamp, usually an NE-51, as an indicator. At maximum permissible recording level, the audio signal fed to the lamp causes it to fire.

Although a single neon lamp can indicate when recording level is too high, it cannot also indicate whether the level is too low. Therefore frequent practice is to use a second lamp which ignites at a lower recording level. When the latter is ignited, the recordist knows that he is putting enough signal on the tape to produce an adequate signal-to-noise ratio. Thus he tries to set the gain control of the recorder so as to keep the second lamp ignited as long as possible without causing the distortion indicating lamp to fire more than once in a while.

Indicators vs Meters

The electronic indicators in common use are generally not as uniform, reliable, or stable as a good meter, particularly the vu meter, which has been especially designed for audio use. Production tolerances for electron-ray tubes or for neon lamps allow individual units to produce significantly different indications. While one NE-51 may require as much as 65 volts a.c. to fire, another may need only 45 volts. This is a difference of about 3 db and when recording at a high level, a further increase of 3 db can raise distortion substantially. When an electron-ray tube or a neon lamp must be replaced, the new one may have significantly different characteristics than the original one.

The skilled recordist knows that maximum permissible recording level varies according to the audio source, since a given amount of distortion is less offensive for certain sounds than others. A meter enables him to adjust recording level fairly precisely. The electron-ray tube permits considerably less accuracy, while the neon lamp is essentially a "go-no-go" device and does not provide intermediate indications.

Electronic indicators require substantially greater driving voltage than do meters. An electron-ray tube such as the 6E5 requires between −6 and −8 volts to completely close the eye, while the neon lamp requires at least 65 volts d.c. and, more typically, about 80 volts d.c. to fire.

Bias current fed to the record head has a fairly critical value if one is to obtain an optimum combination of low distortion, good treble response, and high signal-to-noise ratio. Therefore in tape machines employing a vu meter as a record-level indicator it is frequent practice to also use the meter for measuring bias current by means of a switching arrangement. Unfortunately, the electron-ray tube and neon glow lamp do not have sufficiently accurate characteristics to permit their use in this respect.

Advantages of Indicators

Much lower cost is the primary advantage of the electronic indicator. In terms of function, the electronic indicator has an important advantage in that it reads true peak level, without the lag that occurs in the meter because of the mechanical nature of the latter. The electronic indicator responds immediately to the rapid transients found in music and speech, whereas the relatively slow-moving meter only partially indicates true level. To minimize distortion and maximize signal-to-noise ratio, it is highly desirable to know the actual peak recording level. In the case of the meter, an allowance has to be made for the difference between the meter reading and the actual level on transients. Depending upon the program metered, peaks may be from 5 to 20 db above the meter reading, so that considerable experience is required to accurately judge the true peak level.

Electron-Ray Circuits

A typical magic-eye circuit, employing a 6E5, is shown in Fig. 2. The ray control electrode partially deflects the electron stream flowing from cathode to target, thereby forming a shadow in the circular fluorescent pattern caused by this stream. When the amplifier grid goes negative and the amplifier plate therefore goes positive, the control electrode, connected to the plate, also goes positive, and its ability to deflect (repel) the cathode-target electron stream is decreased, thus the shadow decreases, that is, the eye closes.

The 6E5 grid is driven by an audio voltage from a stage of the recording amplifier. To provide the correct amount of signal to the grid, so that virtually full closure of the eye corresponds to maximum permissible distortion, a voltage divider is employed. This consists of R_1 and R_2 in Fig. 2.

A circuit such as Fig. 2 allows the electron-ray tube to flicker rapidly as it follows the changes in audio signal and is, therefore, difficult to gauge the maximum eye indication. To correct this condition a "floating action" circuit, like that shown in Fig. 3, is often used, maintaining the eye for a short while at the maximum position reached.

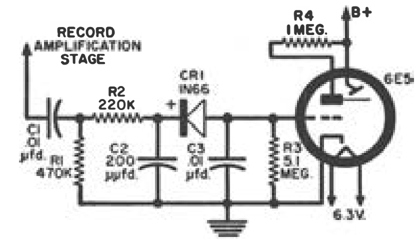


Fig. 3. Typical electron-ray tube circuit uses "floating action." See text.

The values of R_2 and C_3 are chosen so that C_3 is substantially charged by the negative portion of the audio signal in about .001 second through the 1N66 diode. The very high back resistance of the diode requires that C_3 discharge essentially through R_3. The high value of R_3 results in a discharge time of about .05 second for C_3, so that the magic eye reading is maintained long enough to be readily noted.

If it becomes necessary to replace CR_1 in a circuit similar to Fig. 3, the same type of diode must be used. If a popular diode such as the 1N34 were used as a replacement, its much lower back resistance would impair the performance of the circuit. Instead of a crystal diode, some tape recorders employ the spare half of a dual triode—with plate and grid tied to form a diode—in order to achieve the required high back resistance.

Neon Lamp Circuit

Fig. 4 shows the neon lamp in a typical record-level indicator circuit. In some tape machines enough audio voltage is available to fire the lamp. About 65 volts a.c. or 90 volts d.c. is the maximum required to fire the lamp, although appreciably less is often sufficient. Audio voltages of this magnitude are sometimes present as the result of using a high inductance head in the recorder. This, in turn, requires a large constant-current resistor in series with the head (to prevent the head's inductance from discriminating against treble frequencies). Because of the high circuit impedance, a large voltage —65 volts a.c. or more—may be necessary to drive sufficient current through the head to obtain the desired recording level on the tape. By employing a power tube such as a 6AQ5, 6V6, EL84, etc., voltages of the required magnitude can be obtained. The power tube that is used ordinarily serves a double purpose, because in playback it is switched into service as an output tube that is employed to drive the self-contained speaker in the tape recorder.

In many tape machines, however, audio voltages of sufficient level to drive the neon lamp are not available. Therefore the neon lamp is "biased" by a d.c. voltage obtained from the power supply. In this case the audio signal has to supply only the difference between the biasing voltage and the voltage needed to fire the lamp. In Fig. 4, a voltage divider network comprising R_2 and R_3 produces a 58-volt d.c. bias voltage. The audio voltage is in series with the 58 volts d.c. Positive halves of the audio signal raise the total voltage to the magnitude necessary to ignite the lamp.

Fig. 5 is a circuit where a "normal" lamp is added to indicate whether recording is at sufficiently high level. The "normal" lamp fires at a lower audio signal, about 6 db less, than that required to fire the "distort" lamp. The latter requires a higher voltage because of the voltage divider action of R_2 and R_3. Instead of a d.c. biasing voltage, the supersonic bias current is employed here to help fire the lamps; C_2 and C_3 adjust the bias current to the proper level.

The neon lamp has a built-in "floating action" inasmuch as the extinction voltage is appreciably less than the firing voltage. For example, if 90 volts d.c. is necessary to fire the NE-51, the lamp will stay on until the voltage drops to about 50 volts d.c.

Loading Distortion

Irrespective of the type of record-level indicator used, meter or electronic, precautions must be taken so that the indicator will not seriously load down the audio signal and thereby cause significant distortion. This requires that the impedance of the indicator circuit be considerably greater—at least 10 times—than that of the audio circuit impedance. If one is tempted to make changes in a record-level indicator circuit, one must be careful not to substantially reduce the load impedance presented by the indicator.

In a typical electron-ray circuit such as the one shown in Fig. 2, application of the positive portion of the audio signal to the grid causes electron flow from cathode to grid; since the cathode is at ground, no negative bias exists on the grid. In effect, on positive signal

Fig. 4. Typical neon glow lamp circuit.

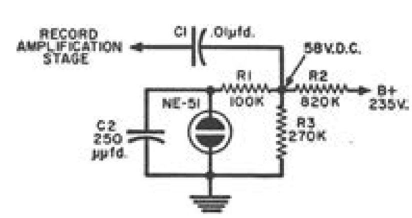
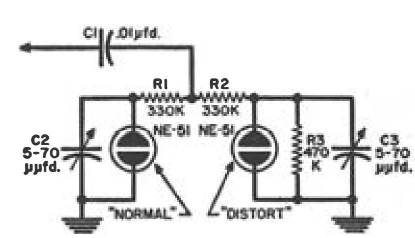


Fig. 5. Circuit using two neon glow lamps.



swings the load (the magic-eye tube) presents a small resistance to the signal in view of the current flowing from cathode to grid. On negative signal swings, however, the effective resistance between cathode and grid becomes infinite since there is no current flow between these tube elements. This changing load resistance would cause severe distortion if it were not for the intervening resistance between the signal and the tube, R_1 and R_2 respectively in Figs. 2 and 3.

A similar thing happens when a neon lamp is employed as an indicator. The effective resistance of the lamp changes from about 100,000 ohms when it is fired to infinity when it is extinguished. Again a series resistor between signal and lamp is necessary to prevent the changing resistance from loading down the audio source. R_1 serves this purpose in Fig. 4, and R_1 and R_2 in the circuit of Fig. 5.

Preventing Bias Pickup

Ordinarily it is of fundamental importance that the bias current supplied to the record head does not reach the record-level indicator, thereby causing the latter to give a false reading. In Fig. 2, bias current is filtered out by C_2 in conjunction with R_1 and R_2. In Fig. 3, filtering is essentially due to C_2 in conjunction with R_2. In Fig. 4, filtering is accomplished by C_2 and R_1. In Fig. 5, we have the exceptional case where bias current is deliberately allowed to reach the neon lamps, as will be further discussed below.

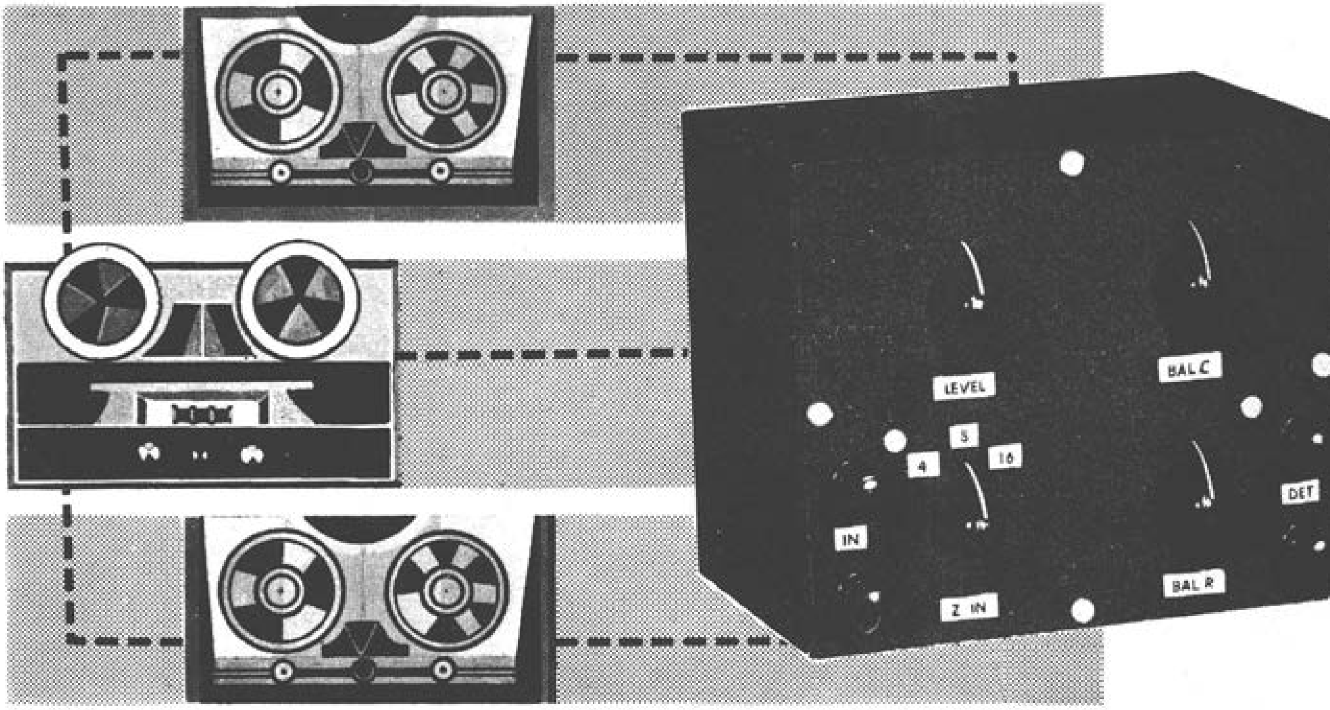
One of the means employed to minimize the amount of bias current reaching the record-level indicator is to pick off the audio signal prior to the stage that drives the record head. Thus the tube that supplies current to the head serves as a buffer between the bias current and the indicator.

Adjustment

If the record-level indicator is not fairly accurate in its indications, serious over-recording or serious under-recording may result. Generally, tape recorders employing an electronic indicator do not provide means for readily adjusting the indicator to correspond to maximum permissible recording level. However, Fig. 5 is an exception. The neon lamps in this circuit are fired by a combination of the audio signal and a certain amount of bias current. The variable capacitors regulate the amount of bias current reaching the lamps and thereby afford a means of adjusting the level at which they ignite.

If adjustments are desired in the circuits of Figs. 2, 3, and 4, one could change values in the voltage-divider networks. In Fig. 2, one could change R_1. In Fig. 3, one could change R_3, although this would also require corresponding changes in C_3 and R_2 to preserve the "floating action" time-constants of the circuit. In Fig. 4, R_2 or R_3 could be changed to place a different "bias" voltage on the lamp.

Measure That Flutter



Flutter tester is built in a 4 x 5 x 6 inch metal box.

By **H. R. WALTER**

Construction details on tape recorder flutter tester that can be made from commercially available parts.

ELECTRONIC service technicians, serious-minded audiophiles, and hi-fi hobbyists are always faced with the problem of tape recorder flutter and are concerned with the best possible mechanical adjustments to keep flutter to a minimum. Flutter is caused by variations in tape speed as the tape passes over the recorder heads and although it may be understood that minimum flutter results when optimum mechanical adjustments are made in the tape transport mechanism, together with replacement of worn parts such as drive wheels, belts, capstans, pressure rollers, etc., the question often arises, how can one know when optimum conditions have been attained?

Tape speed variations, or flutter, are particularly noticeable when a sine-wave signal or sustained notes from a musical instrument, such as the violin, are reproduced from the magnetic tape. Listening tests may result in reduction of flutter as various mechanical adjustments are made, but this method does not provide assurance that speed variations have actually been minimized.

The author has worked with laboratory-type flutter indicators which indicate the per-cent flutter of a mechanical system when used to reproduce a sine-wave signal. Since these instruments provide a high degree of accuracy, they are quite costly and are, therefore, usually found only in laboratories engaged in audio work.

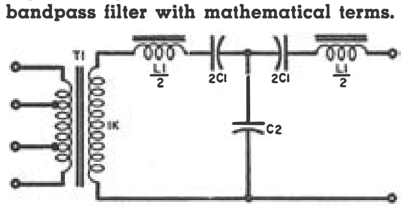
The author has often wondered if a practical instrument, using readily available parts at reasonable cost, could be constructed to enable the audio worker to check the results of mechanical adjustments and provide visual indication of his efforts.

Principles of Operation

A flutter indicator will measure variations in a sine-wave signal frequency, reproduced from a tape transport mechanism, due to speed variations of the magnetic tape.

The input circuit consists of an impedance matching transformer which may work into a bandpass filter network. The primary winding of the transformer may be tapped for common impedances, such as 4-, 6-, 8-, 16- and 600 ohms. The bandpass filter is designed for the flutter indicator fundamental frequency and attenuates hum and noise components in the recorder and amplifier system outside of the passband.

Fig. 1. Diagram of 3-element "T"-section bandpass filter with mathematical terms.



A special form of general purpose, or Wheatstone, bridge is connected to the output of the bandpass filter. This is known as a resonance bridge and is composed of three resistive arms and one series-resonant arm. By manipulating a capacitance-balance and a resistance-balance control the bridge can be balanced at the fundamental frequency and usually about 2% either side of this frequency, that is, minimum indication will be obtained on the null detector since at resonance the inductive and capacitive reactances are equal and, therefore, cancel out, leaving only a resistive component in this series-resonant arm.

With variations in tape speed, the fundamental frequency recorded on the tape will also vary and these deviations from the fundamental will result in a reactive component in the series-resonant arm. Under this condition the bridge, of course, will be unbalanced and the null detector will indicate some value above the minimum reading. Since the bridge becomes unbalanced when the tape test-signal frequency varies with tape speed, the null indicator will furnish information concerning the tape speed variation, or flutter. The null indicator, or meter, may be calibrated in per-cent flutter which is proportional to the frequency deviations. The flutter tester to be described was built to make comparative flutter tests and so was not calibrated. However, some calibration procedures are suggested later in the article.

Circuit Analysis

The input and bandpass filter circuit is shown in Fig. 1. The input transformer, T_1, is a line-to-speaker-voice-coil type and may have a tapped low-impedance winding or a single impedance for the input. This transformer works into the three-element "T" section bandpass filter which was designed for a frequency of 3500 cycles. Mathematical equations used in the filter design are given later for the benefit of the reader interested in constructing a filter with different operating characteristics.

The design frequency of 3500 cycles and impedance of 1000 ohms were used because it made possible the use of a transformer, inductors, and capacitors that happened to be available. Any design frequency may be chosen, but it would seem desirable to work between about 1000 and 4000 cycles. Low frequencies result in large inductors and capacitors. Higher frequencies bring on loss and intercoupling problems and more critical parts placement.

The bridge circuit, as mentioned previously, is a modified form of the Wheatstone bridge, known as a resonance bridge. In this bridge configuration, three of the arms contain only resistive components and one arm includes both inductive and capacitive reactances. See Fig. 2. The bridge is balanced by manipulating potentiometer R_4 and variable capacitor C. The reactance arm is adjusted by the capacitor to obtain a series-resonant condition, causing cancellation of the reactive components, resulting in only a resistance impedance at the audio signal frequency. If the input frequency to the bridge increases or decreases from the fundamental frequency, this will exhibit a reactive component and an a.c. vacuum-tube voltmeter or other null detector placed across the null indicator terminals will give a reading above the balanced condition, depending on the amount of frequency deviation. The null detector may be a v.t.v.m., oscilloscope, or headphones, but for this purpose an a.c. voltmeter is most suitable.

As in the case of the bandpass filter, mathematical treatment for the bridge circuit is given later for the reader wishing to design his own circuit. The complete schematic diagram of the flutter tester with parts values is shown in Fig. 3.

Any physical arrangement of parts will be satisfactory but there are several necessary precautions. The inductors should be of the shielded type. They should not be mounted too close to each other in order to avoid coupling effects. High "Q" inductors are desirable and for best results this is especially important in selecting the bridge circuit inductor.

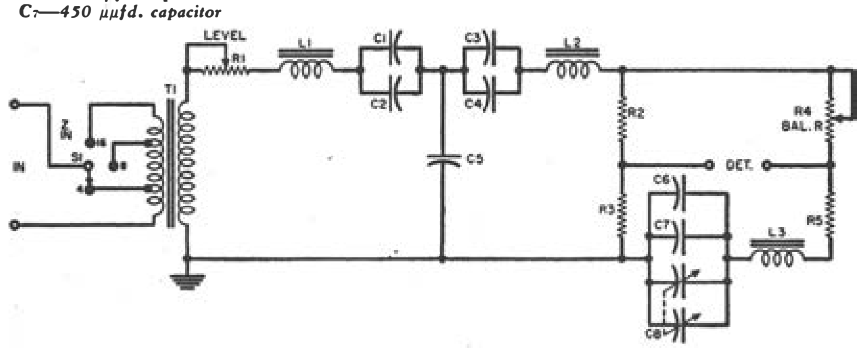
The desired bandpass for the filter network, as designed from the mathematical formulas, was obtained when the response was measured with resistive terminations. The band shifted slightly when the filter was connected in the complete circuit of Fig. 3. These filters should operate with resistive terminations with no, or very little, reactive components. Transformer reactance could possibly be cancelled out, but by experimentally altering the series capacitors in the network slightly, correction can be obtained.

This flutter tester was built for a service technician for making comparative flutter checks, to provide information concerning mechanical adjustments, and to indicate when the best possible adjustment has been affected. For this purpose the following procedure is recommended.

Feed a 3500-cycle signal from an audio oscillator to the input of the tester. Extreme accuracy of the oscillator frequency is not required since the bridge circuit covers the range from 3400 to 3600 cycles. Connect an a.c. vacuum-tube voltmeter across the null detector terminals. The oscillator output control and input control of the

Fig. 3. Flutter tester circuit. Other input Z's, including 600 ohms, may be used.

R_1—500 ohm pot
R_2, R_3—1500 ohm, 1/2 w. res.
R_4—1000 ohm pot
R_5—100 ohm, 1/2 w. res.
C_1, C_2, C_3, C_4—.004 μfd. capacitor (see text)
C_5—.04 μfd. capacitor
C_6—.003 μfd. capacitor
C_7—450 μμfd. capacitor
C_8—365/365 μμfd. miniature var. capacitor (Lafayette MS-142)
L_1, L_2—.3 hy. inductor (UTC Type VIC-9)
L_3—.5 hy. inductor (UTC Type HQB-5)
S_1—S.p. 3-pos. rotary switch
T_1—Line-to-voice-coil trans. 1000 ohms to 4, 8, 16 ohms (Stancor A-8104)



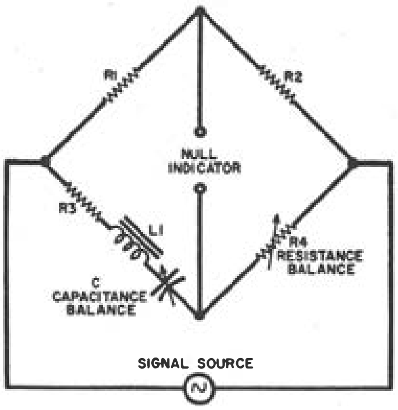


Fig. 2. Modified Wheatstone bridge circuit.

flutter tester should be set for a suitable reading on the meter. Adjust the balance controls on the tester to obtain a minimum reading on the meter. The sensitivity of the voltmeter should be increased by means of its range switch, as required, to obtain the minimum indication. Make a note of this reading. Measure and note the audio voltage input to the tester. The same voltmeter may, of course, be used for this purpose. Now, feed the oscillator to the tape recorder and record this same signal frequency on a tape in the normal manner. This tape is then played back through the recorder and the output fed to the input of the flutter tester. The input impedance of the tester should be selected to be close to the output impedance of the recorder. Adjust the signal level by means of the volume control on the recorder to obtain the same voltage level to the tester that was noted when the oscillator was fed to its input. Observe the reading of the voltmeter connected across the null detector terminals. After making mechanical adjustments or replacing worn parts on the tape transport mechanism, this same test tape is again played back to the flutter tester. The main purpose of the test is to obtain the lowest possible reading on the vacuum-tube voltmeter, or null detector, which corresponds to minimum flutter.

In making this comparative test, it should be noted that the balance controls are adjusted only when originally feeding the oscillator directly to the flutter tester and are not to be changed when playing back the signal recorded on the tape to the tester. The foregoing procedure may at first sound involved, but actually it isn't once the operator runs through the test once or twice.

If the recorder level is not high enough, or the output impedance does not provide a suitable match, the recorder may be played through the regular audio system or through a utility amplifier. The unit shown, however, operated satisfactorily with as little as 0.2 volt to the input.

Other Procedures

Since the flutter tester was constructed for comparative flutter tests, it was not equipped with a self-contained null detection meter nor was it calibrated to indicate the magnitude of the existing flutter, which is usually expressed as a percentage. One method that will give an idea of the per-cent flutter is as follows.

Record a 3500-cycle signal from the oscillator and play back this tape to the flutter tester. Adjust for minimum reading on the null meter. Remove the recorder output from the tester and connect the oscillator in its input. Adjust the oscillator output control to obtain the same signal voltage to the tester as existed when the tape was played. Vary the frequency of the oscillator to again obtain lowest reading of the null meter. Do not change the balance controls. Note the oscillator frequency which gives this null indication. Now we can learn something about the flutter. If the deviation frequency is found to be higher than the fundamental frequency, then, *Per-cent Flutter* $= [(f_2 - f_m)/f_m]\ 100$, where f_m is the mid, or fundamental, frequency and f_2 the deviation frequency. When the deviation frequency is found to be lower than the mid-frequency, then: *Per-cent Flutter* $= [(f_m - f_1)/f_m]\ 100$, where f_1 again represents the deviation frequency.

Consider the following examples. $f_m = 3500$ cycles, $f_2 = 3570$ cycles. Then, $[(3570 - 3500)/3500]\ 100 = 2\%$ flutter. Also, if $f_1 = 3430$ cycles then, $[(3500 - 3430)/3500]\ 100 = 2\%$ flutter. Thus we may term these two results as $\pm 2\%$ flutter which is actually the percentage change from the mid-frequency.

Two per-cent would represent a considerable amount of flutter since present-day recorders have flutter percentages somewhat below 1%. It is interesting to note that the author, when checking flutter as just explained, found one of the earlier model machines had 2% flutter, whereas a late model unit costing $50.00 less than the older unit, had only 0.3% flutter. This indicates the great advances that have been made in tape transport mechanisms. Of course, the old model may have done better with new parts but since they were not badly worn it probably would not have done better than about 1½%.

The flutter tester could be equipped with a self-contained a.c. meter and calibrated in the manner just outlined where the 2% flutter point was found. Meter scale markings would be from 0 to 2%. Accuracy will depend on how the constructor interprets the frequency of his oscillator for each setting.

A d.p.d.t. switch may be used to connect the meter to the bridge null connections or to the input of the bridge to calibrate the meter before making a flutter check. The meter would include a "CAL" marking on its scale and the input level would be adjusted so the meter would deflect to this mark when switched to the bridge input.

A preferred method for calibrating the tester, if the builder is in a position to do so, is to calibrate the device against a laboratory-type flutter indicator.

This section is included for the benefit of the reader interested in designing his own filter and bridge circuits.

The following terms and equations apply to the bandpass network. The term f_1 represents the low cut-off frequency while f_2 is used to indicate the high cut-off frequency. The geometric mean, or mid-frequency, of f_1 and f_2 is f_m. Therefore $f_m = \sqrt{f_1 f_2}$. Bandwidth of the circuit is $(f_2 - f_1)/f_m$. The characteristic impedance is symbolized by R.

The inductance of L_1, in Fig. 1, is then found from the equation $L_1 = R/\pi(f_2 - f_1)$ where f_1 and f_2 are in cycles per second, R is in ohms, and L is in henrys.

The capacitance of C_1, in Fig. 1, is given by $C_1 = (f_2 - f_1)/(4\pi f_1^2 R)$ and the value of C_2 is found by solving the equation $C_2 = 1/\pi(f_1 + f_2)R$ where C is in farads.

In the bandpass filter network of the complete circuit of Fig. 3, the values chosen were $f_m = 3500$ cycles, $f_1 = 3248$ cycles, $f_2 = 3773$ cycles, bandwidth 0.15, and $R = 1000$ ohms.

Mathematical data for the resonance bridge circuit of Fig. 2 is as follows: When the four arms of the bridge are purely resistive, then the bridge can be balanced, when $R_1/R_2 = R_3/R_4$. Also $R_3 = (R_1/R_2)R_4$, where R_3 is the total resistance in this arm, including the series resistance of L. Now the inductive reactance is $X_L = 2\pi fL$ and the capacitive reactance is $X_c = 1/(2\pi fC)$. In Fig. 3, L_3 and the parallel combination of $C_6 - C_7 - C_8$ form the series-resonant circuit and the values of the component parts were chosen so that resonance occurs at the mid-frequency, f_m, previously selected for the bandpass filter with ½ of C_8. At this frequency, $X_L = X_c$, leaving only a resistive component, and the bridge balances. This resonant condition is obtained with the rotor plates of variable capacitor C_8 about half meshed. The capacitor will actually tune the resonant circuit from 3400 to 3600 cycles—which is 2.86% above and below f_m.

TIGHTENING WOBBLY PLIERS

By CARLTON A. CALDWELL

IF THE jaws of your long-nosed pliers become loose so that they wobble noticeably, they can be tightened more nearly like new by expanding the hinge pin. Lay the pliers flat on an anvil of some sort (an axe head should do), and hammer the pin—lightly, with a ball-peen hammer—around its edge a dozen or so times. This will swell the pin a little and make for a better fit. Remember, hard steel is brittle: make sure you are holding the pliers flat on the anvil and don't pound too hard.

Test the pliers occasionally as you proceed to make sure that you are not overdoing the remedy. This would make the pliers too hard to open. If you work gradually, you may get the pin just a little too tight, at worst. If this should happen, all you need is a drop of oil around the pin. That, and working the pliers back and forth to loosen them, should restore the joint to the right tension for proper use.

By HERMAN BURSTEIN & HENRY C. POLLAK
Co-Authors, "Elements of Tape Recorder Circuits"

Reducing Tape Recorder NOISE and HUM

AT reasonable levels of reproduction, the electronic components in any high-fidelity system deserving of the name produce virtually no audible noise or hum. However, an important exception often occurs in the case of the home tape recorder. It can still be said of relatively few moderate-price tape machines that they exhibit low enough noise and hum levels to qualify as "high-fidelity" instruments.

This does not imply gross negligence on the part of tape machine manufacturers. Rather, it reflects the difficulties that must be coped with in tape playback. The signal produced by the playback head is seldom more than 5 millivolts, with 2 or 3 millivolts maximum being more usual. Compare this with the 20 or more millivolts obtained on peaks from the popular magnetic cartridges, and it is easy to see why tape amplifiers have a special problem. Furthermore, the signal produced by the tape playback head must undergo tremendous equalization, consisting of 36 db bass boost if the NARTB standard is followed. By comparison, the output of a magnetic cartridge undergoes only 20 db of bass boost. Substantial treble cut is applied to the signal from the magnetic cartridge—at the same time reducing noise—whereas the signal from the tape playback head undergoes no such de-emphasis and, in fact, may receive some treble boost instead to compensate for gap-width loss.

All-in-all, keeping noise and hum in a tape machine down to a level compatible with high-fidelity standards is no small feat. The following checklist seeks to remind or inform the technician or technically inclined audiophile of ways to combat noise and hum. Quite likely, a tape machine that at one time was sufficiently quiet is so no longer. Or it may be possible to improve a machine that was always noisy. Or in constructing a tape amplifier it may be possible to guard against the many pitfalls that await the builder. The measures to be described are not necessarily effective in all circumstances, but at one time or another they can prove worthwhile.

Reducing Noise

1. Tube Type: Tubes have been developed with special low-noise characteristics for critical applications. Such are the EF86, Z729, ECC83, 12AY7, 5879, etc. A low-noise tube can sometimes be directly substituted, without socket, wiring, or circuit value changes. Thus the ECC83 and 12AY7 can be used instead of the common 12AX7; these are twin triodes. Sometimes a new socket may be required as, for example, if one replaces a 6AU6 with an EF86 or Z729; these are pentodes. When replacing a 5879 with an EF86, the socket and circuit values are the same, but the connections to the socket must be rewired.

2. Tube Selection: Low-noise tubes command premium prices, yet this is not complete assurance that one is getting a satisfactory tube. Although the chances of getting a good tube are greater with premium than with conventional types, success can be assured only by being able to select the best of several units of a given type.

3. Resistor Noise: Resistor noise shows up in the very high gain circuit of a tape playback amplifier. Low-noise resistors should be used in at least the first stage and, preferably, in the first two stages. In subsequent stages, a conventional 2-watt resistor can be used for noise reduction. Low-noise types should be used not only in the plate circuit but also in the cathode circuit if the cathode resistor is unbypassed. Wirewound, non-inductive resistors are best, but most costly; moreover, the resistor does have some residual inductance and is therefore susceptible to hum pickup. Deposited metal film resistors can be virtually as noise-free as the wirewounds. One should be wary of the deposited carbon resistor in this application. Although some deposited carbons have excellent noise properties, others are little better than ordinary resistors.

4. Tape Hiss: The substantial treble boost in playback, employed by a number of tape machines, accentuates tape hiss caused by imperfect cancellation of the minute magnetic fields on the tape, called "domains." A tape machine conforming to NARTB equalization principles applies all or most of the necessary treble boost during recording instead. To change a tape amplifier to NARTB equalization in both recording and playback modes is fairly major surgery. Tape hiss can perhaps be kept down by subjecting the tape to a bulk eraser instead of always relying on the erase head. Tape hiss may be accentuated by significant departures from flat treble response in the control amplifier, power amplifier, or speaker. The speaker is the most likely culprit. In the case of commercially recorded tapes, substantial hiss may be recorded on the tape as the result of too many "generations" of tape.

5. Head Noise: Heads tend to gradually become magnetized due to the asymmetrical audio waveforms presented to them; the asymmetry, in effect, constitutes a d.c. component. A magnetized head has a d.c. field which records a d.c. pattern on the tape that varies with the irregularities in the coating and base of the tape. In playback, these irregularities are heard as noise (modulation noise), therefore it is advisable to demagnetize the heads periodically, say after 10 to 20 hours of use, with an electromagnet designed for this purpose. Heads may also be magnetized by accidental contact with a magnetized tool, such as a screwdriver, or by d.c. current flowing through them because of a leaky coupling capacitor.

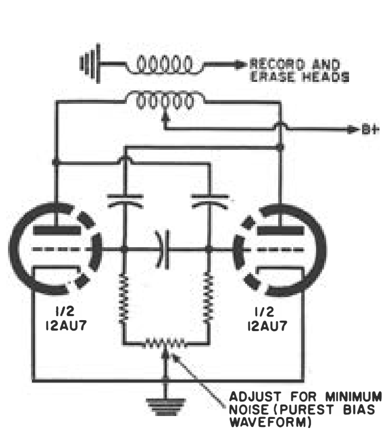
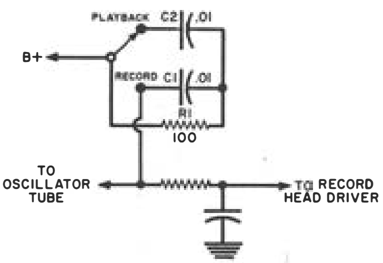


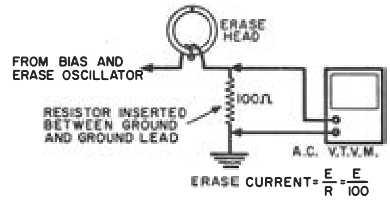
Fig. 1. Push-pull bias oscillator.



NOTE: C1 OR C2 IN SERIES WITH R1 PREVENT CURRENT SURGES BY PROVIDING PATH FOR BRIEF CONTINUED CURRENT FLOW DURING SWITCHING

Fig. 2. Method of preventing surges.

Fig. 3. Measuring the erase current.



NOTE: IF V.T.V.M. DOES NOT HAVE SUFFICIENT SENSITIVITY, TRY A 1K INSTEAD OF 100Ω RESISTOR

6. Bias Waveform: The waveform produced by the bias oscillator and fed to the record and erase heads should be as close as possible to a pure sine wave. Distortion in the waveform can be a source of noise because such distortion may represent a d.c. component, magnetizing the head and causing noise, as already explained. The bias waveform should be viewed with a scope to pinpoint gross distortion, if present. But this is only a preliminary check, because much less than the approximately 5% distortion discernible on a scope can produce appreciable noise, therefore voltages and components in the oscillator circuit should be checked. Possibly the oscillator is overloaded by a defective erase head or other defective component. High quality recorders generally employ a push-pull oscillator, typically a 12AU7 or 12BH7, and in the case of poor waveform the circuit should be checked for matched components, namely grid resistors and coupling capacitors. Fig. 1 shows a typical push-pull oscillator circuit. This one also incorporates a means of balancing each half of the oscillator for maximum symmetry and, therefore, purest waveform; that is, the grids are connected to ground through a balancing potentiometer.

7. Tube Voltages: At times tube noise can be significantly lowered by operating the heater at moderately reduced voltage (through a series resistor), for example, about 5.5 volts in the case of a 6.3-volt tube. Noise also tends to vary inversely with plate current so that a higher "B+" voltage supply or smaller plate resistor can reduce noise. (But in the case of triodes, reducing the value of the plate resistor may raise distortion; in the case of pentodes, it will significantly reduce gain.)

8. Shock-Mounting: Even the best of tubes are somewhat susceptible to microphonics, set up by vibration of the transport mechanism. Shock mounting is therefore advisable for the first tube and possibly for others as well. The socket may be spaced away from the chassis by using rubber grommets on the socket-mounting screws.

9. Switching Transients: Clicks may be recorded on the tape when motors, relays, etc. are switched on or off. This can be prevented by placing capacitors —suitable values are about .01 to .05 μfd.—across the switches and relay contacts.

10. Current Surges: Sudden application or removal of "B+" from the oscillator and record-head driver stages, which is apt to occur when switching between record and playback, may cause clicks and pops to be recorded. Also, these current surges may cause the heads to become magnetized, producing noise as previously explained. Again, capacitors can solve the problem. Fig. 2 is a typical circuit, incorporating capacitors C_1 and C_2 in series with a resistor R_1, so that "B+" will be gradually applied or removed from the record-head driver and oscillator.

11. Imperfect Erasure: If the tape has previously been recorded and is imperfectly erased when next used for recording, some of the previous program material will be audible, often annoyingly so. The fault may lie in an erase head of poor design or one with shorted turns. Insufficient erase current may be reaching the erase head. This can be checked as shown in Fig. 3, where current is determined by Ohm's Law by reading voltage across a resistor, namely $I = E/R$. The amount of current should be that specified by the manufacturer, with 10 to 20 ma. a typical value for erase heads in most home tape machines. If erase current is too low, then the oscillator circuit and the components coupling it to the erase and record heads must be checked. Even though erase current is of the value suggested by the manufacturer, the erase head may be ineffective because the current *frequency* is too high, so that the winding capacitance of the head acts as a substantial short-circuit. Thus for a given amount of erase current, an erase head may erase well at 65 kc. and poorly at 100 kc. When a tape has been recorded at an excessively high level, even a normally operating erase head may be unable to achieve adequate erasure. An electro-magnetic bulk eraser is then required, provided one is willing to erase the entire tape and not merely one track. If one track must be kept, the only recourse is to put the tape through the record process, with the volume control all the way down, thereby subjecting one track to a preliminary erasure. Of course this is very time-consuming.

12. Print-Through: Also known as pre-echo and post-echo, this refers to the appearance (on a given portion of the tape) of the signal on adjacent layers of tape. Print-through is apt to become audible if the tape has been recorded at high level; also if the tape is stored in a warm place and/or near magnetic fields. For a given set of circumstances, print-through is greater for the thinner tapes.

13. Record Level: Excessive noise may simply be due to a low recording level. This can be checked by comparing the level of a tape recorded on one's own machine with a commercially recorded tape that sounds clean (some recorded tapes have obvious distortion due to over-recording). Inadequate record level may be due to malfunction of the record level circuit, which should be checked.

14. Gain Control: If the gain control is located in an early stage in the playback circuit, turning down the control will fail to simultaneously reduce the noise of the later stages. It may be advisable to install a dual gain control with one section in a later stage.

Reducing Hum

1. Tube Type and Selection: As with noise, use of a preferred type of tube and selection of the best tube out of several of the same kind can lead to a rather substantial amount of hum reduction.

2. Tube Shield: Tube shields should be employed for at least the first stage and preferably the first two. As an extra precaution, the shields should be demagnetized by means of the bulk eraser used for erasing tapes. Extra shielding can be provided for the tube by wrapping it in a layer of "Co-Netic" *(Perfection Mica Co.)*, an effective, but expensive, shielding material.

3. Tube Demagnetization: Sometimes hum can be reduced by demagnetizing the input tube (and others as well), using the bulk eraser. Since the eraser is usually very powerful, caution should be exercised so that the tube is not brought near enough to the electromagnet to dislocate the tube elements themselves.

4. Hum-Bucking Pot: One of the simplest, least expensive, and most effective means of reducing hum is to use a hum-bucking pot in the heater supply, as shown in Fig. 4A. In a number of inexpensive tape machines, a.c. is employed on the heaters, with the electrical center of the transformer's heater winding connected to ground. However, electrical center is not necessarily the best grounding point for maximum hum reduction.

5. Biasing the Hum-Bucking Pot: Sometimes the effectiveness of the hum-bucking pot can be increased by connecting the arm to about 20-50 volts "B+," as shown in Fig. 4B, instead of to ground. A large capacitor, of about 20 to 40 μfd., should be employed for a.c. grounding. By making the heater positive with respect to cathode, this prevents a.c. current flow from heater to cathode due to emission of electrons by the heater.

6. D.C. Heater Supply: A d.c. heater supply can make the choice of an input tube and the problems of lead dress less critical. Fig. 5 shows a circuit for converting an a.c. 6.3-volt heater supply into a d.c. supply for either 6- or 12-volt tubes.

7. Rectifier Tube Hum: The rectifier tube develops a magnetic field which may extend far enough to cause hum pickup by another component. Thus it would be poor practice to put the rectifier and input tubes next to each other on the chassis.

8. Head Shield: A major source of hum pickup is the playback head. Professional tape machines usually surround the heads with a heavy shield during operation, leaving a gap just wide enough for the tape to pass through. Home recorders ordinarily employ less expensive and generally less effective head-shielding measures.

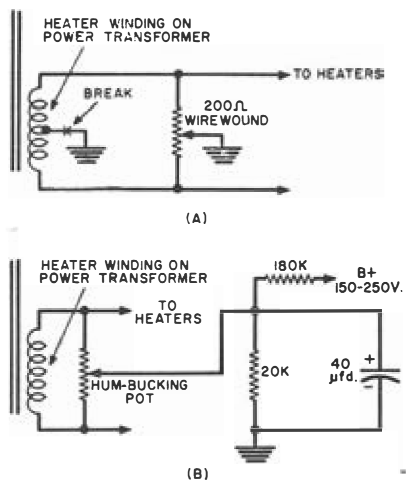


Fig. 4. Installing a hum-bucking pot.

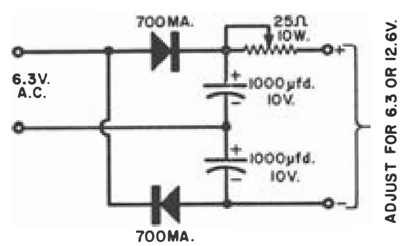
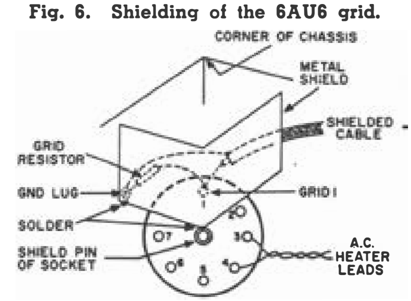


Fig. 5. Rectifiers are rated at 18 v. (Use Federal's type 1013 or equiv.)

Fig. 6. Shielding of the 6AU6 grid.



Sometimes hum can be reduced by placing a small piece of shielding material, such as "Mumetal," "Co-Netic," or a piece of silicon steel from a junked power transformer, in front of the head gap.

9. Input Grid Shield: Significant hum reduction, especially if a.c. is used on the heaters, is possible if the grid of the input tube is shielded, as shown in Fig. 6. The shield can be formed from a piece of tin can, soldered to the center pin of the tube socket.

10. Input Lead: The lead from the playback head to the input of the first playback tube is apt to be relatively long and thus susceptible to hum pick-up. It is advisable that this lead be shielded, but using low capacitance co-axial cable to minimize high-frequency losses.

11. Lead Dress: Grid and plate leads must be carefully routed away from hum sources, such as heater leads, power leads, power transformers, etc. It is advisable to dress the grid and plate leads as well as associated components, namely resistors and capacitors, close to the chassis, which acts as a shield.

12. Chassis Ground: The chassis should not be employed as a ground return because magnetically induced hum currents circulating through the chassis may be coupled to low-level audio signals. Instead, one common ground point, well-soldered to the chassis near the grid of the first playback tube, should be used.

13. Ground Loop: An inadvertent ground loop, that is, a multiple path to ground, may cause substantial hum. For example, if the ground lead of the playback head is returned to ground through the shield of a coaxial cable, if this shielded wire accidentally contacts the chassis, and if the chassis is separately grounded, then there are two paths to ground for the head.

14. Location of Power Transformer: The power transformer should be mounted as far as possible from the playback head.

15. Mounting the Power Transformer: In building a tape amplifier, it is preferable to use a transformer that mounts above chassis rather than flush with it (part above and part below), because in the case of flush mounting the chassis acts to extend the transformer's magnetic field.

16. Orientation of Transformer and Motors: Hum may be alleviated by rotating the transformer and/or motors, as mounting conditions permit.

17. Shielding the Transformer and Motors: Well-shielded motors and transformers are considerably more expensive than the garden variety and thus not too likely to be found in moderate-priced tape machines. Yet the power transformer and the transport mechanism motor(s) are very potent sources of hum, most likely picked up by the playback head. If the tape machine is troubled by hum from these sources, one may try wrapping a shield around the transformer and motors.

18. Type of Chassis: In building a tape amplifier, an aluminum chassis is preferable to a steel one, for the former does not radiate hum. Also, it has less resistance to circulating currents, including those produced by hum fields, so that the resultant hum voltages present in the chassis are lower.

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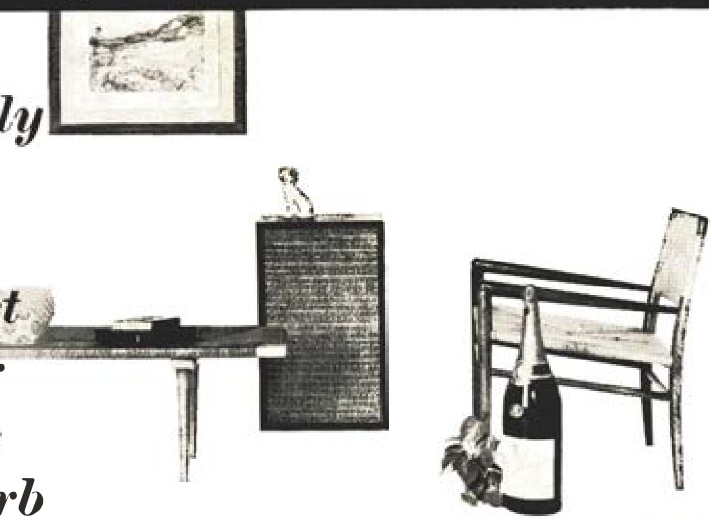

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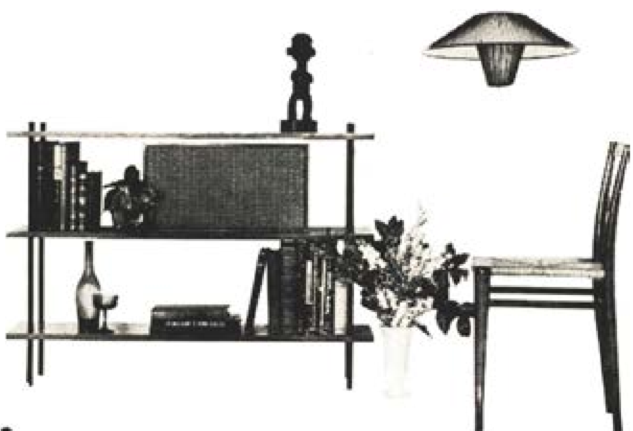

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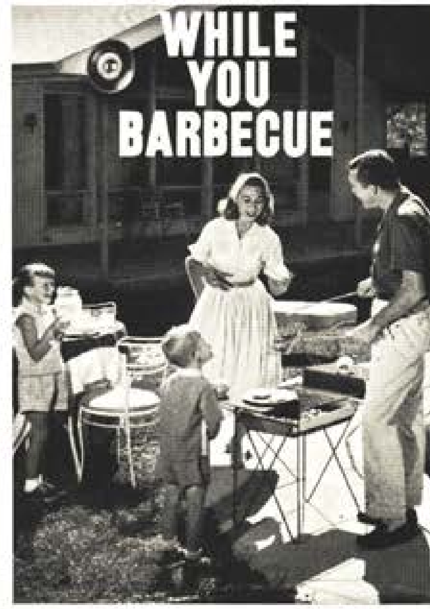

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