# A New Stereophonic Amplifier*

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*Summary*—A central feature of the new design of a stereo amplifier is an output transformer with original features that makes possible reduced cost and improved performance at the same time.

This paper discusses a varied possibility of design objectives for a stereo system, and explains the way in which the new output transformer functions. By variation in its method of use, or in choice of parameters, a whole range of amplifiers can apply advantages in different proportions or degrees.

The basic design of an output transformer, which is essentially inexpensive to make, provides for separation between "left" and "right" as well as crossover, and combining networks for mixed lows, if desired, without additional external circuits. It makes possible a new type of tone control, achieving high performance economically, using feedback, and/or improved matching between amplifier and loudspeakers over the entire frequency range as well as better separation and efficiency than the single-ended and push-pull transformer matrix can give.

One particular amplifier is discussed in detail, while a more general discussion shows possible application to more diverse design objectives.

## Design Objectives

While the primary objective behind this development was economy—making it possible to achieve stereo of reasonable quality at considerably reduced cost—it was also agreed that realistic standards of quality must be met; perhaps it would be possible to achieve superior quality at lower cost. This can often happen, where a simplified, more logical approach replaces an older, more complex one. It has happened in the group of systems to be described in this paper. While there is one central feature that characterizes each amplifier of the group developed, its application is so flexible that a whole range of amplifiers has been designed to suit the entire range needs of a phonograph line.

The most expensive part of any audio amplifier has always been the output transformer. Some economy was effected in an earlier development[1] that used a circuit configuration similar to single-channel push-pull, but carried one stereo channel in each side of the push and pull. This also effected some economy in output transformer by utilizing a closed-cored, high-efficiency unit for the push-pull, or monophonic element, with a gapped, lower-efficiency unit for the single-ended, or stereo element.

The new development carries this economy much further. Efficiency is improved because the current for each channel has only to flow through one primary winding and one secondary winding, where the two-transformer matrix uses two windings in series for both primary and secondary (Fig. 1). Also the maintenance of good separation is less dependent on precise control of the number of turns in the various windings.

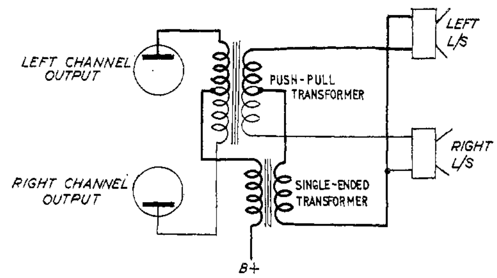


Fig. 1—The two-transformer matrix limits the economy that can be achieved because the signal current path for each channel flows through a winding on both transformers in series, both primary and secondary; current path for one channel is shown by heavy lines.

## Requirements

In deciding what is acceptable performance, the question of separation and how well it is maintained at extreme frequencies, low and high, must be considered.[2] The character of separation is also important: most measurements of separation do not determine what the signal is that leaks from one channel to the other; it is assumed to be the left program leaking into the right channel, or vice versa; what is more important is the leakage of distortion components from one channel to the other.

To illustrate this difference: if the leakage between channels is pure program, some 12 to 15 db is probably adequate for almost all purposes; improvement beyond this would not noticeably improve the stereo effect. We set a minimum of 20 db as a target to insure a good margin. But if the leakage consists of distortion components, then 20-db crosstalk represents 10 per cent distortion!

Separation, at the low frequencies particularly, resolves into two kinds from the practical program viewpoint. The test method usually employed assumes the signal is present in only one channel, and must not be allowed to leak into the other. The practical program, except the type that has been "doctored for super

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[1] B. B. Bauer, *et al.*, "A two-way stereophonic amplifier," *Audio*, vol. 42, pp. 19–20, 92; October, 1958.

[2] W. H. Beaubien and H. B. Moore, "Perception of the stereophonic effect as a function of frequency," *J. Audio Engrg. Soc.*, vol. 8, pp. 76–86; April, 1960; also in IRE Trans. on Audio, vol. AU-8, pp. 144–153; September–October, 1960.

stereo," has some of the signal present in both left and right channels. Aural separation is achieved by differences in the intensity and phase with which the signal corresponding to different program sources or instruments is contained in the two channels.

The conventional method of testing for separation merely determines by how much a system will degrade the proper amount of intensity separation. For example, if a piece of program has the intensity in one channel of 14 db above that in the other, and the system separation is 20 db: the leakage represented by signal content is 1/5th, and that added by the system (assumed in phase) is 1/10th, making a total of 3/10ths; so the 20-db system separation may degrade a program separation of 14 db to only 10 db.

But program can differ in timing, or phase, as well as in intensity, between the two channels. In a normally recorded program, both differences contribute at lower frequencies because sound intensity throughout a studio will not vary appreciably at these frequencies, and timing can only change by a fraction of a period from point to point.

The effect of incorrect timing can well be illustrated by reversing the phase of one channel at the loudspeaker. On some programs, all sense of stereo location is lost, but an exaggerated sense of spaciousness remains. In others, where a different recording technique was used, there may be little difference. But, however the program is recorded, the reversing phase of one speaker always gives an impression of bass deficiency. The only case where this may not occur is in the "super-stereo" type program, if all the bass is in one channel. This indicates that in most program material, both correct phase or time relationship and intensity differences are important at these frequencies.

Of course, coupling between channels will degrade phase as well as intensity difference, but in a different way. The distinction is mentioned here because it can explain some of the discrepancies between experimental tests conducted to determine the need for maintaining separation at the extreme low and high frequencies.

## The Output Transformer

A new departure in output transformer design forms the central feature of the new line of amplifiers, with different aspects of its flexible range of attributes utilized for different applications. To understand the functioning of this transformer, it will be necessary to explain the properties of the quantity called leakage inductance, and we can illustrate the development of the new unit in terms of earlier applications.

## Leakage Inductance

For filter design, leakage inductance combines properties of iron-cored and air-cored inductances. An air-cored inductance uses a very long magnetic path in air, linking with the coil [Fig. 2(a)]. An iron-cored inductance occupies this path with magnetic material of high permeability [Fig. 2(b)]. This greatly increases the $Q$ of the inductor, but also introduces a nonlinear component, due to the nonlinear relationship between $B$ and $H$ in the magnetic core.

Leakage inductance is usually defined as a measure of the unshared magnetic flux due to imperfect coupling between the coils and is quantitatively dependent on the primary flux that does not link the secondary, and vice versa. While this is a correct definition, the concept of it as imperfect coupling leads to the notion that leakage flux is a small percentage of the main flux, and possesses the same nonlinearities the main flux does which is not true.

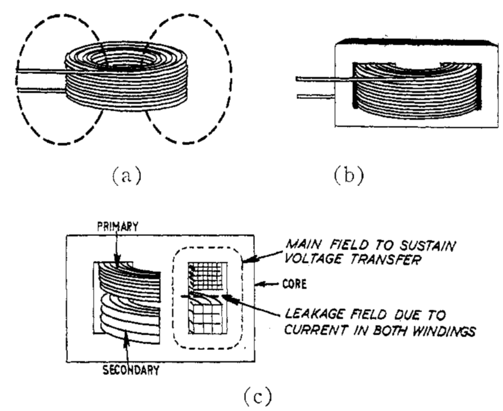


Fig. 2—Comparison of properties of (a) air cored, (b) iron cored and (c) leakage inductances, for use in filter design; the leakage inductance arrangement is shown in part section with the paths of main and leakage field indicated.

An alternative way to define leakage inductance avoids this difficulty. All the flux that stays in the core couples both primary and secondary in most designs. (In special designs magnetic material may be inserted in the leakage flux path, but these will not be considered in the present paper.) The leakage flux is induced to pass between the coils [Fig. 2(c)] by the combined current in both windings (principally due to load current drawn from the secondary and reflected to the primary) and is responsible for a voltage difference that is added vectorially to the voltage that would be induced in the windings by the main flux. So leakage inductance is the inductance between two coils occupying the same core due to a magnetic path between them, which path, usually, is wholly in air. Leakage flux is only such when it leaves the surrounding core. From this understanding, the properties of leakage inductance may be seen to differ from conventional inductance in two respects.

1) Although it has the properties of an inductance, it provides circuit isolation because two coils are involved, which may or may not be connected externally. The inductance is defined in terms of the

voltage induced in both coils, referred to the turns in one of them, by the rate of current change in both coils.

2) It has a $Q$ considerably superior to an air-cored coil because the magnetic path length, although in air, is considerably reduced. At the same time, it does not, in itself, have the distortion-generating property of an iron-cored inductor. The leakage inductance is entirely due to magnetic path *in air*. The part of the magnetic path within the core material is not active in the leakage inductance element, but is part of the magnetizing current characteristic of the transformer of which leakage inductance is another element.

The magnitude of leakage inductance can be varied in several ways. In considering its magnitude, reference must be made to a specific winding. The same leakage inductance will have two distinct values, if the windings between which it appears do not have the same number of turns. Leakage inductance is proportional to the square of the number of turns in the reference winding and is governed also by the geometry of the windings.

Increasing turn length, the spacing between windings, or the dimension of the windings transverse to the leakage path, will increase leakage inductance [Fig. 3(a)]. Increasing the dimensions parallel to the leakage path, or multiplying the number of winding sections by division made parallel with the leakage path [Fig. 3(b) and (c)] will reduce leakage inductance.

An early use of leakage inductance for audio filter design yielded an efficient and compact low-pass filter, using the parameters generally known as $m$ derived.[3] Later, the same principle was extended to produce a transformer with built-in crossover, for feeding low- and high-frequency loudspeakers with their respective ranges of frequency.[4] This use was the first to demonstrate a way of achieving high-pass as well as low-pass action through the use of leakage inductance (Fig. 4).

Fig. 4(a) represents a condition where the capacitor in shunt with the output winding operates in conjunction with the leakage inductance, which is effectively in series with the the transmission path to provide low-pass action. In Fig. 4(b), the position of $L$ and $C$ are transposed. This is achieved by using windings with the same numbers of turns, phased so the output voltage can be the drop across the leakage inductance between the two windings.

When stereo first became popular, the use of two winding assemblies on the same core as a means of economy was envisaged.[5] In this case, the leakage inductance between windings disposed on the two limbs of a core-type construction would provide potential separation between channels, and by correct phasing of the signal in the two channels, single-ended outputs could be used in such a way that the core loop has no resultant polarized (dc) magnetization.

[3] N. H. Crowhurst, "Leakage inductance," *Electronic Engrg.*, vol. 21, pp. 129–134; April, 1949.
[4] N. H. Crowhurst (Tannoy, Ltd.), "Improvements Relating to Electrical Transformers," Brit. Patent No. 734,346; July, 1955.
[5] N. H. Crowhurst, "Stereophonic Sound," John F. Rider Publisher, Inc., New York, N. Y., 1st ed.; 1957.

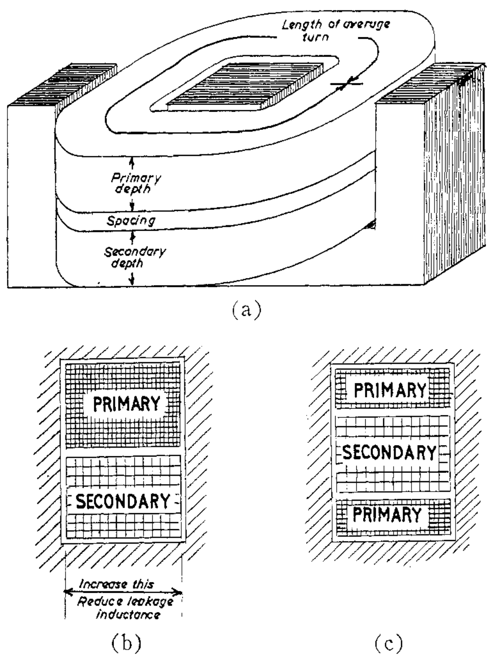


Fig. 3—Dependence of leakage inductance on the geometry of the transformer into which it is built: (a) dimensions shown here will each increase leakage inductance in proportion to themselves; (b) inverse proportion on this dimension; (c) the first simple step in sectionalizing to reduce leakage inductance.

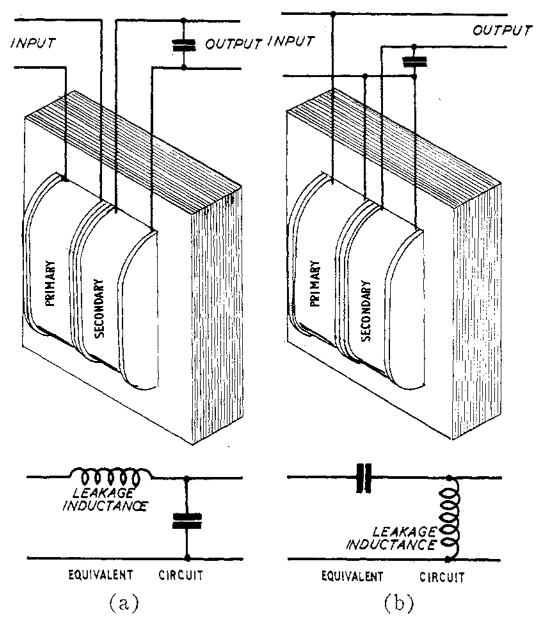


Fig. 4—Connections with double-wound transformer to utilize leakage inductance in (a) low-pass and (b) high-pass filter. Note that in (a) no electrical connection is necessary between input and output, while in (b) such connection is necessary and the turns in the windings must be equal.

### The New Development

In a sense, the basic element of the new design combines the features represented in the literature.[4,5] But it does more than this; it gains some advantages due to the particular method of combination that do not pertain to either feature individually.

The basic configuration for the new transformer is shown in Fig. 5. The outputs of a push-pull stereo amplifier are accommodated on symmetrically disposed windings of a core-type transformer. Without the third winding, which incorporates the crossover function, the separation progressively improves as frequency rises due to the rising impedance of the leakage inductance between the two winding assemblies.

Basically, the third winding provides low-pass filter action between a mixture from both the main windings, which can be used to feed a common-bass unit. A capacitor connected across this third winding completes a two-element low-pass filter action. A full equivalent circuit of this joint leakage inductance action cannot conveniently be drawn with conventional circuit elements, nor can a simple equivalent circuit be constructed. The leakage inductance as measured from either main winding to the third winding, in conjunction with the leakage inductance as measured between the two main windings, do not conform to either loop or mesh configuration of a three-terminal network.

Below the crossover frequency, which is determined by the referred leakage inductance value for the turns used in conjunction with the capacitor, chosen to synthesize a filter having constant resistance properties, most of the energy is tightly coupled to the third winding. Above crossover, progressively less of the energy is coupled to this winding. But what there is, is strictly "sum" or monophonic signal.

The secondary windings on the main winding assemblies have the same number of turns as one half of the third winding. By appropriate phasing, the output voltages oppose at low frequencies, so that the left- and right-speaker terminals receive less and less signal below crossover. Above crossover, the bypassing action of the capacitor across the third winding, in conjunction with the leakage inductance between each main winding and the third winding, serves to couple the left and right speakers directly to their respective windings.

Immediately above crossover, up to a frequency where the leakage inductance between the left and right windings is directly adequate to provide separation, the presence of the third winding also serves to improve separation. The sum signal appearing across it is split in two and subtracted from the partially-coupled signals appearing across the main windings. The result is that each output is much more fully separated from the other (Fig. 6).

It should be noted that the main windings are not, in themselves, treble windings, although when separate common bass is used, their main function is to supply load current at treble frequencies. But the voltage across each main-winding secondary is tightly coupled to its respective primary winding. By grounding one end of each main-winding secondary, and the center tap of the third winding, the filter action, both low pass and high pass, can be achieved, while the main windings give full-range voltage for feedback purposes.

Notice that the entire load current for the common-bass output is delivered by the third winding. Except in the region just above crossover, the current for left and right is delivered directly from the main windings, although the third winding is in series. The capacitor across it bypasses the higher-frequency currents from the main windings, so voltage drop due to third-winding resistance is avoided. The only currents that cause voltage drop in the third winding are those below crossover, and this is then the active winding, feeding the common-bass speaker. Thus, efficiency can be maintained more easily than with the two-transformer matrix.

The circuit is more efficient, as well as costing less than either two separate transformers or a two-trans-

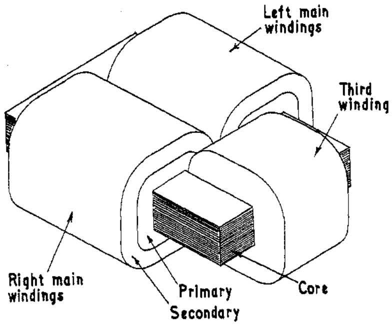


Fig. 5—Physical configuration of the single-assembly transformer.

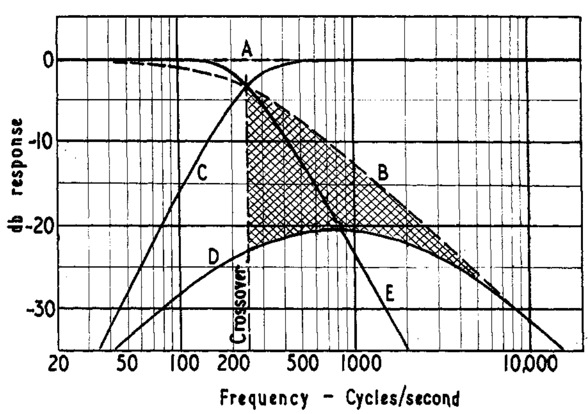


Fig. 6—Effect of the third winding and its capacitor on separation: A (dashed line), response in active channel without use of third winding; B (dashed line), response into inactive channel (breakthrough or crosstalk) without use of third winding; C (solid line), response into active channel with third winding and high-pass filter action; D, response into inactive channel (breakthrough or crosstalk); E, response into common-bass circuit.

former matrix, followed by separate LC low-pass filters to provide common bass. The common-bass filtering is provided in the magnetic circuit without resistive loss additional to normal transformer operation.

## Spurious Effects

It proved important to avoid any internal resonance in the third winding of the transformer above the critically loaded one used for crossover. An early attempt used a simple winding with a center tap. Such a winding has a leakage inductance between its halves, which is very much smaller than the leakage inductance from the other windings (Fig. 7).

However the capacitor is connected—even if two separate capacitors are used to bypass each half separately—a resonance occurs between the leakage inductance from one half, regarded as the exciting winding, and the other, across which the capacitor appears as a virtual shunt load. This is a series resonant circuit that builds up a peak, injected into the opposite channel in the region of 12 kc. The disadvantage of this is that at 6 kc a signal from one single-ended amplifier contains a 12 kc distortion component that resonates in the opposite neutralizing winding to inject a high-amplitude, double-frequency cross-talk component.

The remedy was simple—bifilar winding with opposite ends of the two sections connected together to form the center tap. This makes the coupling so tight that no resonance occurs within the audio range, and where the new resonance might occur, the $Q$ has deteriorated to much less than unity.

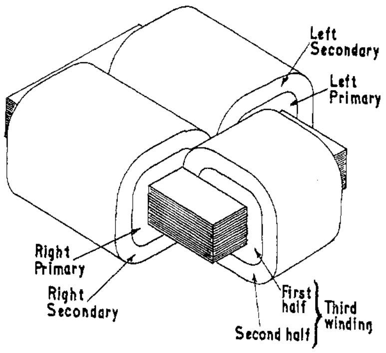


Fig. 7—When the third winding is center-tapped in conventional manner (shown here physically), the leakage inductance between its halves is considerably smaller than that from either main winding to the third winding.

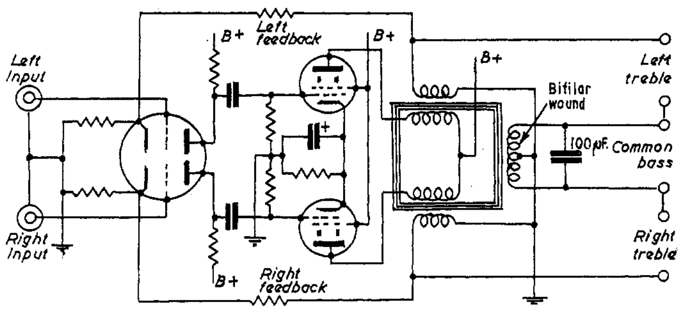


Fig. 8—Schematic of amplifier utilizing the single-assembly output transformer to provide common bass, left- and right-treble outputs, and full-range feedback to drive stage.

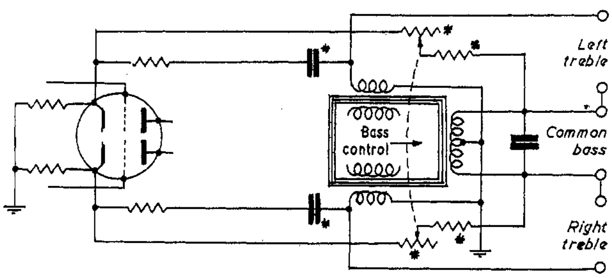


Fig. 9—Partial schematic to show the six extra components needed to add a very effective two-channel-bass control to the circuit of Fig. 8.

## The Amplifier

Variations of the transformer have already been applied in several different amplifiers, but one application in particular shows how this unique circuit can produce advantages in several directions, some of which were not envisaged at the outset of this project.

First developed was a straight amplifier, to which straight feedback was applied, yielding lower distortion and improved separation (Fig. 8). This amplifier yielded enough gain to give full output with only one voltage stage, using a ceramic pickup, and with sufficient margin to allow for turning up the wick on weak records, or to accommodate the people who do not think it's loud unless it's distorted.

## Tone Control

To provide tone-control facilities would require either an extra stage to provide for the loss needed to obtain boosts as well as cuts, or the insertion of reverse-type controls in the over-all feedback. Economy suggested trial of the latter method.

But inverting a bass tone control, to be terminated by the input-stage cathode resistor, is not easy. Then it was realized that the transformer contains its own bass filter, already used for the common-bass output. By augmenting or attenuating the feedback obtained from the appropriate half of the third winding, and attenuating the low-frequency feedback from the main winding by using a suitable value of capacitor in series with its feedback resistor, we have a full-bass tone control (Fig. 9).

This is where the circuit gives a very useful bonus. The low-pass filter that separates bass from mid-range uses a two-reactance filter, enabling a sharper slope boost and cut than is possible with conventional circuits using only one reactance (usually a capacitor) for this purpose. This enables the level of frequencies below crossover (chosen as 250 cycles, which is the center of the musical scale) to be varied quite drastically without noticeable change in gain above this point in mid-range. The tone-control action is superior to many circuits that cost very much more in terms of components and gain.

To complete the tone control, using a similar method for separating the higher frequencies, first a maximum lift is inserted by connecting a resistor and capacitor across the cathode resistor (Fig. 10). Then a high-pass resistor and capacitor pick off frequencies above the point where the lift commences, to provide the adjustable element of feedback above this point. A simple variable resistance, bypassing the fixed one for mid-range gain, completes the treble tone control.

The resulting amplifier produces frequency response, distortion and separation characteristics that represent considerable improvement over its predecessors in a comparable price range; it also has extremely effective tone controls, without the need for extra stages, and because of the fewer components, it effects quite a cost-saving too. The new development makes it easy to improve performance and cut cost at the same time.

## Other Variations

This is only one of a group of amplifiers developed, using the same central principle, with variations. A less expensive version omitted the third common-bass winding, but retained feedback, using it for volume control function including compensation that makes it effectively a loudness control. This produces lower distortion and better bass than was possible in this cost bracket previously, without any increase in cost.

For the really high-quality applications, many variations are possible. If the tone control function is separated to its more conventional location in a preamplifier a greater amount of feedback is used in the power amplifier section to reduce distortion almost to vanishing point. If the common-bass feature is not desired, a change in connections enables two full-range loudspeakers to be used (Fig. 11).

There are other possible advantages of the new transformer and its associated circuitry. For example, by using the common-bass winding in reversed polarity, with full-range units, the impedance matching from the tubes can be made to suit the impedance characteristic of a loudspeaker with its rising value in the bass (Fig. 12).

Above crossover, the main windings only provide separated left and right outputs. Below crossover, the fairly sudden coupling of the third winding adds turns in series with each output so that the impedance match is for a higher value. This enables an amplifier to deliver greater power at the low frequencies where it is sometimes needed without sacrificing the damping that feedback can give, or limiting the properly matched power available for the mid-range and higher frequencies.

Use of this development is not confined to tubes or to a single-ended operation. It may provide its most effective economy to single-ended circuits, but it retains its other advantages, including economic, with circuits using two separate channels with push-pull outputs, or using transistors instead of tubes.

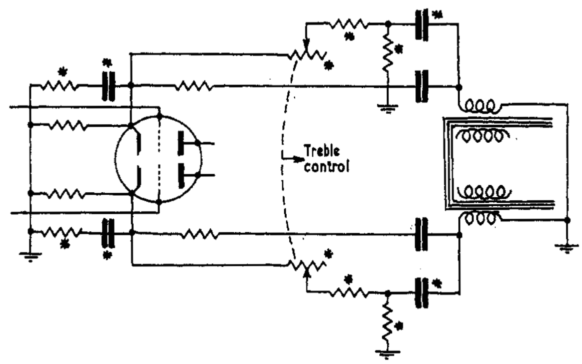


Fig. 10—Circuit of a two-channel treble control, using the additional components marked with asterisks; this circuit is partial, added to the basic circuits of Figs. 8 and 9.

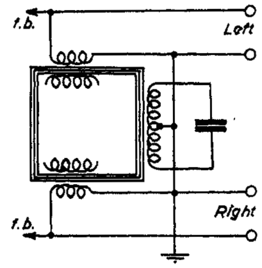


Fig. 11—Change in output connections needed when two full-range speakers are to be used instead of common bass.

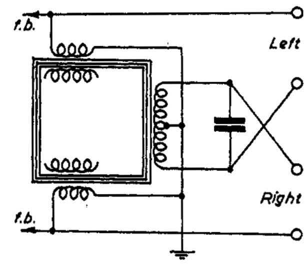


Fig. 12—A further change in connections that can be used to improve matching of the amplifier to the loudspeaker impedance over the whole frequency range.

## Conclusion

The new transformer principle which forms the heart of the new development provides a number of advantages, which may be used in various combinations or degrees in individual amplifier designs:

1) An efficient means of combining mixed lows and retaining separate left and right at frequencies above a crossover built into the design of the transformer.
2) Improved separation for the degree of circuit complexity or precision involved, both in the range immediately above crossover and the extreme high frequencies.
3) The provision of convenient take-off points which may be used for full-range feedback, or for the inclusion of feedback bass tone control, with superior performance; the addition of circuit to include treble control is relatively simple.

4) Where the design is applied to other than a mixed lows system, the built-in crossover may be used to provide more efficient power matching to dynamic loudspeakers at both bass and treble frequencies.
5) With any or all of the foregoing advantages, considerable economy in application because of the extent to which "free" elements within the single transformer are utilized.


Acknowledgment

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# An Improvement in Simulated Three-Channel Stereo*

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*Summary*—**Some two-channel stereo systems have employed a third full-range speaker system in the center, reproducing an equal in-phase mixture of the signals in the two channels. Advantages of this arrangement over the usual two-speaker array are better reproduction of the location and size of central sound sources. A disadvantage is the sizeable reduction in the apparent spread, or distance between flanking sources.**

**The reasons for these effects are discussed, and it is indicated that this disadvantage can be largely overcome by electrically reducing the ratio of sum to difference of the two channels, which ratio was effectively increased by the addition of the center speaker. It is shown that the signals to the three speakers may be regarded as three independent channels with certain signal-to-crosstalk ratios, which are derived as a function of the level of the center speaker and the amount of electrical reduction of the sum-to-difference ratio. The choice of optimum parameter values and appropriate circuits is discussed.**

A METHOD has been devised for improving the geometric fidelity of reproduction of two-channel stereo. In the sections that follow, the defects of present systems are described and it is shown how these may be partly overcome.

## Defects of Two-Speaker Stereo

There is no doubt that a conventional two-speaker system can provide good reproduction of the location and, to a certain extent, the size of sound sources. In order to accomplish this, however, the listener must be satisfied with a narrow speaker spacing and a precisely centered listening position.

If the speakers are far enough apart to subtend a desirably large angle at the listener, and the listener is on the center line or axis of the system, central sources appear larger and their location more indistinct than similar sources at the extreme left or right. Whereas this phenomenon may be aggravated by poor microphoning, it cannot be blamed entirely on the microphone technique, for it occurs even when the technique is such that these central sources are recorded purely monophonically, *i.e.*, with identical signals in both channels. The phenomenon is probably caused by at least the three following factors:

1) *Effect of head rotation:* When a small sound source is directly in front of an observer, and the observer rotates his head slightly to the left, the sound strikes the right ear more perpendicularly and grows louder in that ear, particularly at high frequencies.[1] Sound reaches the left ear more obliquely than before, and its loudness in that ear, therefore, decreases. If the rotation is sufficient, the left ear is shadowed by the head and the consequent attenuation at that ear is even greater. Also, sound reaches the right ear sooner than the left. The observer uses the relation between rotation and amplitude and time differences to pinpoint the source and determine its size. Obviously, if the source is very large, or consists of two widely separated points emitting identical sounds, a given rotation of the head produces less amplitude and time differences at the ears than it would if the source were a single point. The

* Received by the PGA, April 5, 1961.
† Warwick Manufacturing Co., Chicago, Ill.

[1] F. A. Firestone, "The phase difference and amplitude ratio at the ears due to a source of pure tone," *J. Acoust. Soc. Am.*, vol. 2, pp. 260–270; October, 1930.