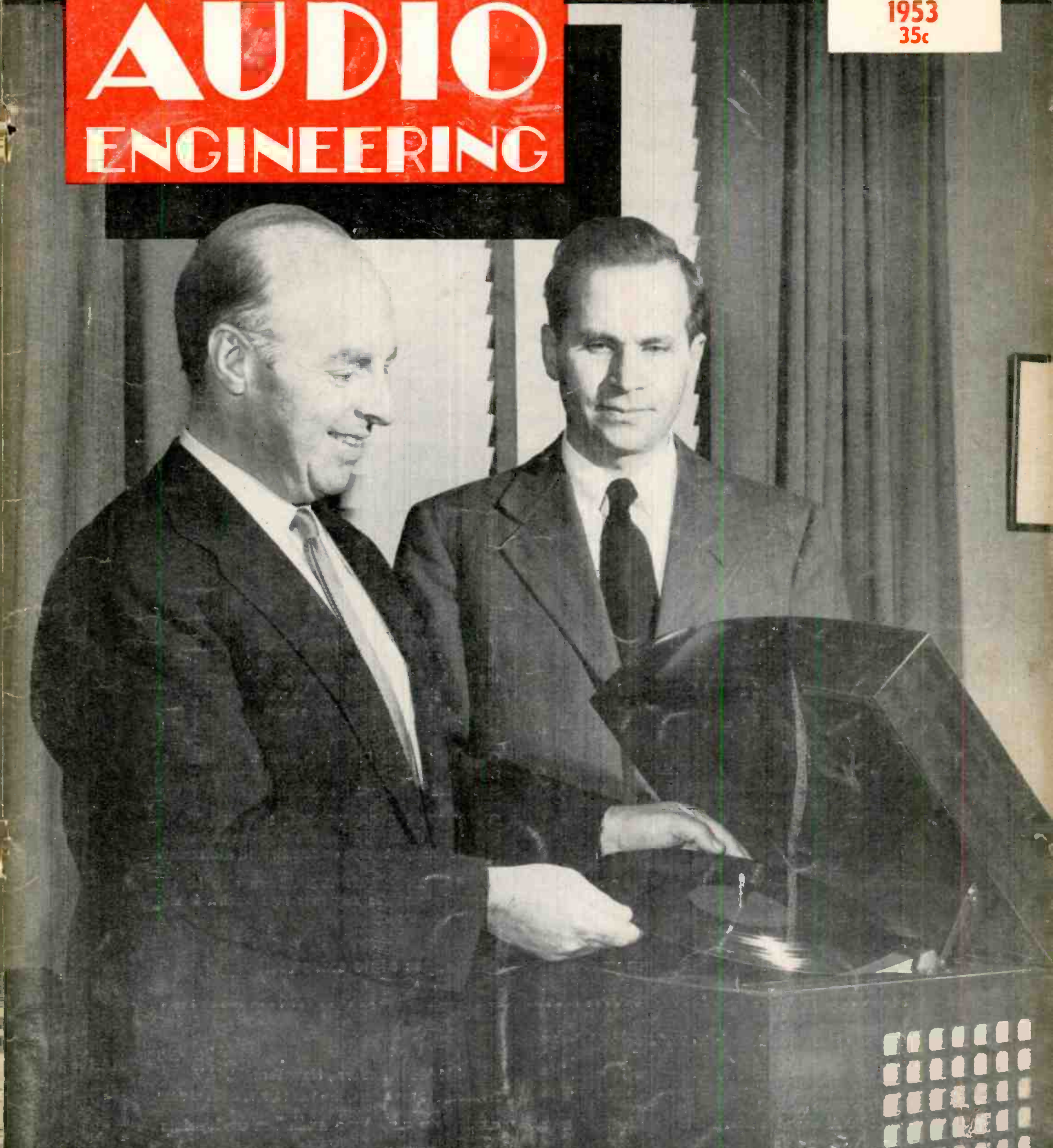
BINAURAL or STEREOPHONIC? See page 22

AUDIO ENGINEERING

JANUARY
1953
35c



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CONTENTS

JANUARY, 1953 Vol. 37, No. 1

COVER

Snowstorms in July are commonplace compared with pictures of table-model phonographs on the cover of Æ. They are no more newsworthy than this one, however. Developed by Dr. Peter Goldmark (right), for Columbia Records, Inc., the instrument offers a standard of audio performance never before approached in a unit of comparable size. At the left is Andre Kostelanetz, celebrated conductor, whose musical ear Dr. Goldmark borrowed more than frequently in designing the equipment.

RADIO MAGAZINES, INC., P. O. BOX 629, MINEOLA, N. Y.

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AUDIO PATENTS

RICHARD H. DORF*

MAGNETIC RECORDING has certainly come a long way since Poulsen's original methods, and most of the progress has been made in the last very few years. One of the problems in obtaining good fidelity with wire and tape is the non-linearity of the B-H curve of the medium. That was solved by Poulsen (and until a relatively short time ago) by superimposing a d.c. bias on the sound. But more recently a.c. bias has become standard.

The field is still new enough, however, from the standpoint of concentrated development and commercialization, to allow the thought that perhaps a.c. supersonic bias may not be the very best bias method. Lawrence H. Connell of Detroit, Mich., proposes a new and unfamiliar system which deserves a good look, at least by those interested in top-quality reproduction. Without experimentation the writer is unable to even begin to pass judgment on it, but Connell's idea is at least well worth examination. It is explained in his patent No. 2,604,546.

Figure 3 shows a magnetization curve which might represent that of a typical recording medium such as the tape commonly in use. For various values of magnetizing force H exhibited by the head, the curve shows values of magnetization B left on the tape. Assuming the tape to be initially unmagnetized, applied values of H between a and a' produce magnetization values between b and b'. Values of H greater than that produce proportionally much larger magnetization. Obviously the steps in the curve will produce distortion if nothing is done.

The distortion is commonly removed by applying a steady a.c. bias which effectively eliminates the step and places the entire recording on the linear portions of the curve. A.c. bias has, however, some effect on higher frequencies; it either fails to record them or erases them to some extent. It is common knowledge that the correct bias value is a compromise between low distortion and good frequency response, the failings in the latter respect usually being made up if possible by additional equalization. D.c. bias, of course, allows use of only half of the available linear portion of the curve and the signal-to-noise ratio is relatively poor.

* 255 W. 84th St., New York 24, N. Y.

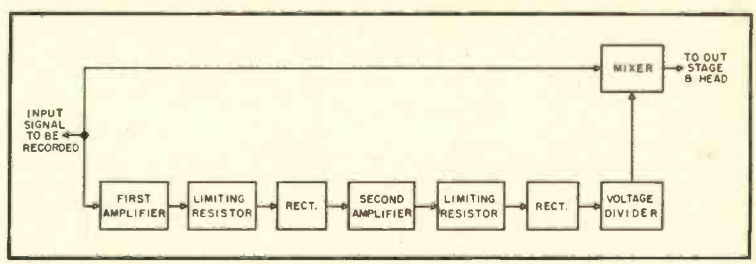

Figure 1.

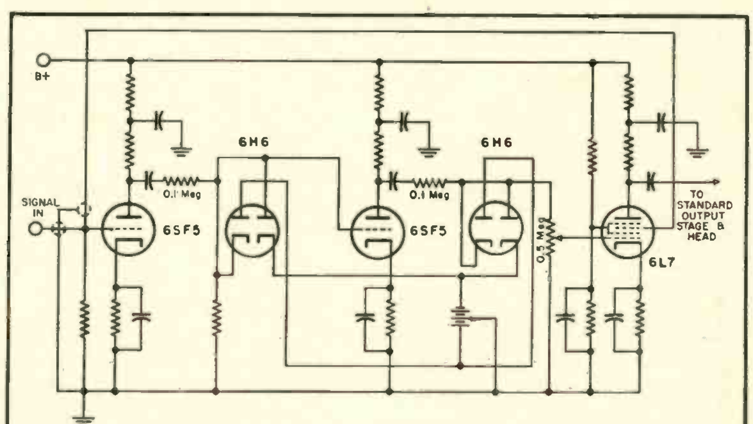

Figure 2.

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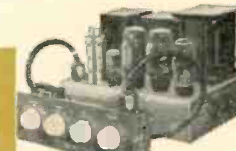

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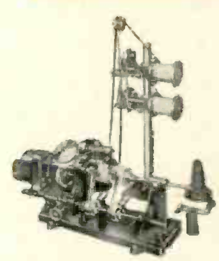

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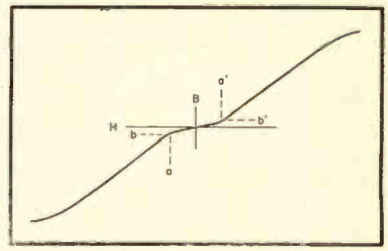


Figure 3.

According to Connell's method, a voltage is added to the signal before it reaches the head. This voltage always has the same instantaneous polarity as the signal voltage and a constant value equal to the distance from the baseline to the beginning of the linear portion of the curve.

Figure 4 shows what this is like. At (A) we see a typical audio wave during the course of a program. All the sounds being recorded are, of course, resolved electrically into a single complex waveform, which is shown.

At (B) there appears the waveform of the specially added voltage, corresponding in phase or polarity to that of the signal in each portion of the signal cycle, but having a fixed peak value. (C) shows the two mixed. The audio signal appears above and below the bracketed dotted lines. The audio never reaches zero. In effect the audio waveforms are plotted with two baselines, one for positive and one for negative alternations. The real baseline—equivalent to zero value of H in *Fig.* 3—is not reached except by the superimposed voltage in passing through it to change polarity.

The wave at (C) is applied to the recording head and thence to the tape as magnetizing force H. The audio modulation is always beyond the *a-a'* points of the H values; thus the magnetization of the tape is always beyond the *b-b'* points or on the linear portions. In this way the audio modulation on the tape is split into two separate but linear portions. The idea seems a little like drinking some bitter medicine in the middle of a meal so fast in relation to your normal eating speed that, practically speaking, the bitter taste did not exist.

The block diagram of *Fig.* 1 shows how the job is done. Signal is applied to one grid of a mixer stage and thence to the output stage and recording head in the usual way. It is also applied to a special clipper network. The first amplifier is a high-gain stage feeding a biased rectifier through a limiting resistor. The top-flattened signal goes through a second high-

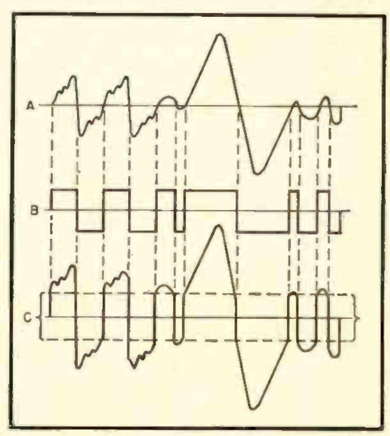
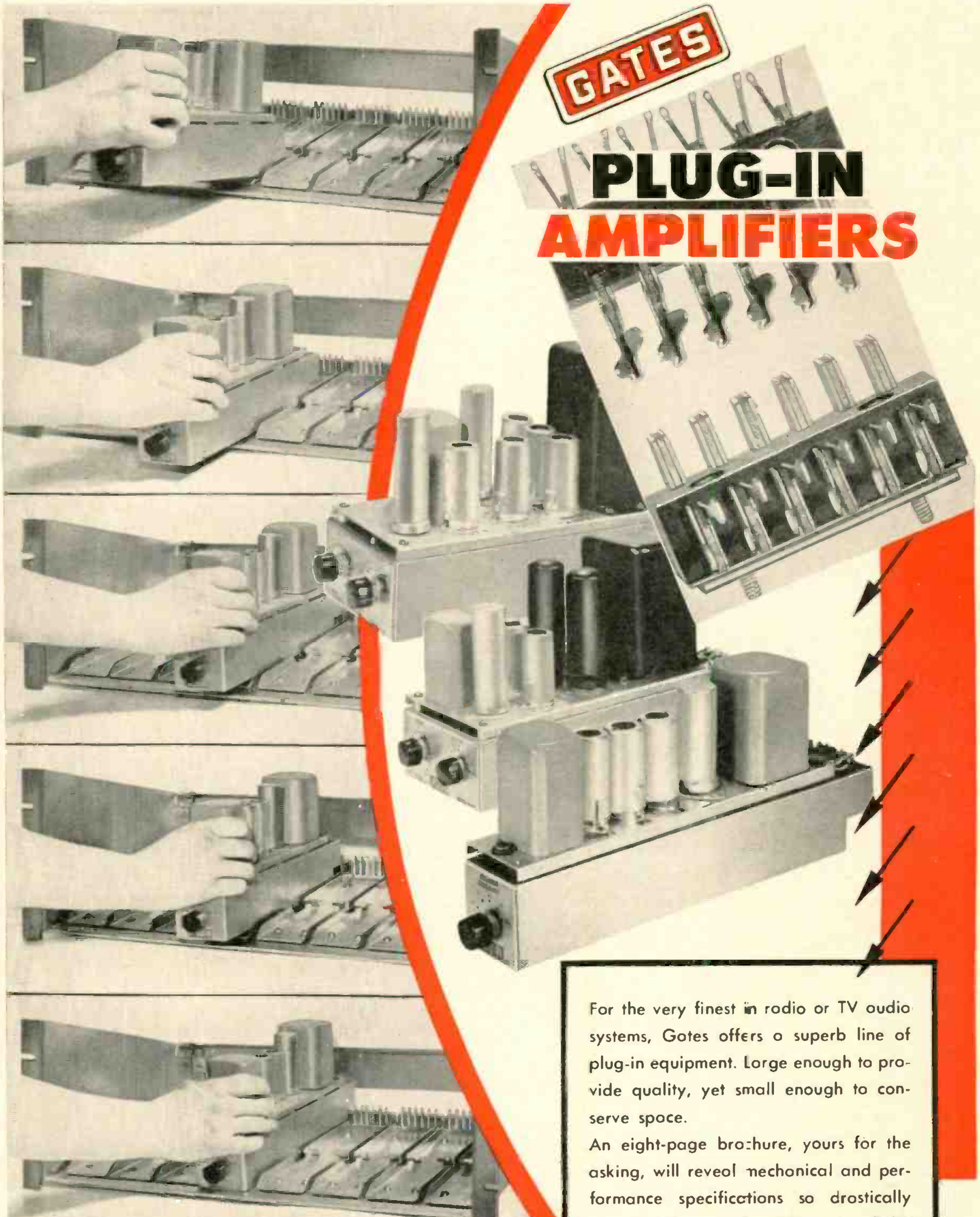


Figure 4.



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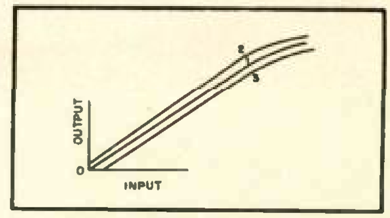

Figure 5.

gain amplifier, limiting resistor, and rectifier, followed by a voltage divider, from which it emerges as a substantially square pulse. Because of circuit arrangement it has the same phase as the original signal and because of the high amplification and clipping its peak voltage is practically constant. This is shown in (B) of *Fig. 4*.

It is mixed with the original audio signal in the mixer tube and the resultant is fed to an output stage and head in the normal way.

Figure 2 shows a circuit arrangement for the purpose. Input signal goes to the grid of a 6SF5 and also directly to grid No. 3 of a 6L7 mixer. The 6SF5's and 6H6's are arranged as indicated in *Fig.* 1, the clipper system ending in the 0.5 meg potentiometer, the arm of which is connected to grid No. 1 of the 6L7. The potentiometer is used to control the level of the "bias" voltage so that it just a little more than eliminates the hysteresis step in the *B-H* curve.

The 6H6's do all the clipping. They are biased by the battery or some other d.c. source so that they always clip at the same value. The 6SF5's are operated as non-distorting amplifiers so that even for a very small audio input signal the desired "bias" pulse will be created at the output.

The system may be calibrated for optimum "bias" value with a vacuum-tube voltmeter connected to each of the two 6L7 signal grids. A random adjustment of the voltage divider is made, with a constant tone being fed to the system. Then a recording is made with various values of signal voltage, the levels of each being read

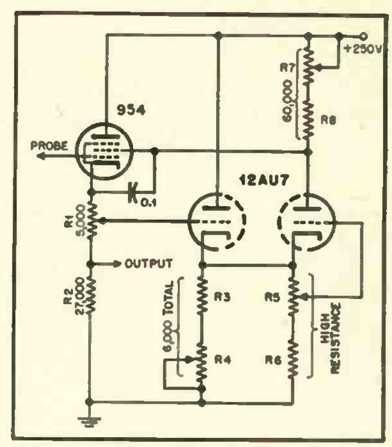

Figure 6.

on the meter connected to grid No. 3. The recording is played back and a curve is made to show audio input *vs.* audio output. The lower portion of the curve should have a relatively straight portion like those in *Fig.* 5. The object is to place the curve so that extrapolation of the straight portion will make it pass through the zero point as does curve 1. If the 0.5-meg potentiometer is set too high the results will be as in

[*Continued on page 51*]

LETTERS

Why Stereophonic or Binaural Reproduction?

Sir:

The most sought-after characteristic of music reproducing systems today is realism, which means the congruence of reproduced music with the original in all of its ramifications, some of which are loudness, frequency response, and spatial distribution. There is the assumption that such a congruence is both desirable and necessary. Why?

The purpose of any musical device is to produce sounds which are pleasing. It so happens that violins, pianos, clarinets, etc., when competently manipulated by composer and artist can be made to produce pleasant effects which we call music. There is nothing sacred about these instruments—they are merely the means to an end. They are generally unwieldy means, requiring infinite patience and practice to achieve pleasant results. They are tolerated only because the music can be obtained in no better way. There are many inflexible individuals who ascribe a certain sanctity to things as they are; unable to understand real or potential changes for the better. Witness the complaints that the harpsichord should not be amplified because it is essentially a weak instrument, and things-as-they-are would lose their intrinsic balance. Since there is no immediately obvious way of making it louder of itself, then it must be forbidden to do so electronically. Also can be cited the question of opera in English; because the librettist wrote for an Italian or French audience, it has been anathema to translate freely into the language which can be understood. At the same time, there is no need to go overboard for change just for the sake of change. We want to change only if there is an advantage in doing so.

The fact that musical instruments must be of finite size to produce their sounds makes it imperative that an assemblage of instruments occupy a space in proportion to their number. As a result of this "accident," any number of methods have been devised to group the members of the aggregation in a compact and aurally balanced manner. The louder percussion instruments are to the rear; the weaker instruments are usually to the front; the arrangement is often amphitheater style. "Chamber" groups are not spread out on a stage, even if it is available. They cluster together, often facing one another, and just far enough apart not to interfere with each other's movements. Desirable auditoriums are those in which reflective but non-selective reinforcement occurs—the sound floods over the audience from the walls and ceiling.

A disagreement of long standing has been that of mike placement—close-to or remote. The argument for remote placement is that a mike in the hall would "hear" the music the same way as a listener in the same location—as though it were necessary to hear the music like a live attendee. If the acoustics were ideal, he might hear the music in a fairly satisfying manner With several mikes scattered throughout the orchestra, and with balance then maintained electronically, the most annoying consequences of instrument size are obviated. The instruments no longer have to be spread out. The result can be ideal in that each instrument would effectively be right under the baton, and its loudness would effectively be the proper loudness. Another advantage is that instruments previously discarded because of their weakness could be reinstated. The harpsichord and the recorder—to mention two which gave way to the piano and flute despite the very rich tones of the originals—are not very appropriate instruments for "live" performances. The ratio of listeners to "canned" music to those of live music being as large as it is (and even larger when the incredible number of dilettantes are weeded from the latter) makes it apparent that the needs of the home listener should come first. Fortunately, some of the recording companies are recognizing this need.

Spatial (binaural and stereophonic) reproduction must come in for some questioning if these views are accepted. Theater organs are enormous structures, and have to be split up, parts on opposite ends of the stage. Because of this, must the home listener be subjected to the same kind of disparity? It is true that spatial systems will increase the realism by recreating, to some extent, the original spatial distribution of the source. Does this aspect of realism enhance the enjoyment of music? To accomplish this with the advocated stereophonic devices there must be a symmetrical room with fairly complete damping all around. The listener has little freedom of movement of sitting position and speaker placement is rather critical. Echoes and reverberation will destroy the effect. Only the echoes of the original room are desired. For some reason the echoes which occur in the home are dirty and much to be avoided. They reduce what remains of spatial effect after concert hall reverberation has done its work.

It is admitted that stereophonic or binaural systems are ingenious. But because a system is ingenious, represents creative achievement, and works well is not sufficient reason for its adop-

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tion. We all know high-fidelity people who play all kinds of peculiar sound effects and bizarre music but who never, by their behavior, indicate any interest in music.

We know, too, of the frustration occasioned by the perplexed layman who is just unable to be impressed with the differences between his console and our super-four-way corner monoliths.

Much is to be said for a spatial system, however, in the case of radio drama. One of the great superiorities of TV plays over those on radio is the localization information of the former. Realism is desired and can be achieved by reintroducing the spatial dimension. Footsteps move across the room, a car pulls up at the left, a door opens on the right, and so on. However, there is little market for canned drama, and the likelihood of two-channel radio networks is slim.

There are numerous "reasons" for coaxial mounting of woofers and tweeters—the foremost being the desire to avoid a split source and interference. Despite the fact that ordinary homes are so live that the effect of standing waves at least equals these unwanted effects from the speaker, these are fairly important considerations. Few people are brave enough to mount woofer at one end of the room and tweeter at the other in an attempt to achieve a certain degree of spatiality. But when we put two speakers in the room, each fed from a different channel, where do all our worries about disparate sources and interference go? It cannot be claimed that the two speakers do not interfere because they are emitting something different. This would be much more true for the woofer-tweeter combination than for the spatial system. The truth is that localization is accomplished because of subtle intensity and phase differences at the ears, and unless the over-all system (including reverberation) is arranged so these special conditions are maintained, the results will either be of separated sources, or of an essentially diffuse and undifferentiable source.

The diffuse source approximates more closely the concert hall which has good acoustics than does the point source, and perhaps more so than does the two-channel system. The diffuse system—one which employs two or more speakers on one channel—is obviously simpler, cheaper, and not limited in program material as is the spatial system which relies on two complete and independent systems, special tape and tape playback mechanisms, or special discs and tone arms. Is it uncomfortable to recall the reluctant realization that music so often sounds better in other parts of the house (despite all the concern about tweeter beam-width) than in the same room with the speaker?

JOHN VERSACE,
5621 Georgia Ave. N.W.,
Washington 11, D. C.

(Even with a few parts eliminated to fit it in this space, this letter is long. However, the thinking seems to be worth passing along to all of Æ's readers. ED.)

Other Reactions

Sir:

While I have not seen the following idea mentioned previously, it may only be that it is too obvious to some.

Are we not trying to achieve a goal which is just about at the end of the rainbow? The job a hi-fi system is asked to do—i.e., to make the listener feel that either the original sound source is in his living room or that he is actually at the location of the original sound source—is just about psychologically impossible. Try-

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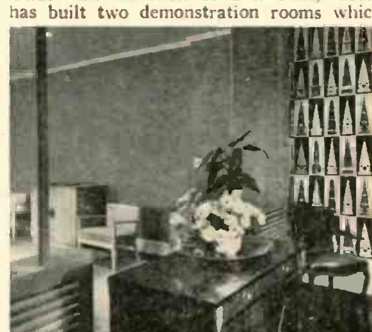
ing to make a listener believe something he knows is not so is difficult, whereas if he believes that what he is going to hear is real, a very poor reproducing system is all that is necessary to uphold the illusion. For example, vocalists with the best dance orchestras nearly always use relatively poor p.a. systems, yet it never dawns on a high percentage of the listeners at the dance that they do not hear the real live vocalist at all. The original source is nearly always completely masked by the reproducing system. I am convinced that if an audience were assembled in an auditorium in which high-fidelity speakers were placed along the front of the stage and an orchestra placed behind an opaque but acoustically transparent screen to produce the music unknown to the audience, a high percentage of this audience would report that the high-fidelity system is good but still does not quite sound real.

I do not mean to imply that we should stop trying to improve high-fi systems, but merely make the suggestion that perhaps we should re-evaluate the state of the art in this light. Maybe some who are becoming neurotic trying to make their systems sound real will be somewhat consoled. Most of the present-day systems sound fairly good to me, but no matter how good they sound, I cannot sit in my living room, close my eyes and be transferred by magic carpet to the symphony auditorium, nor does the magic carpet take me back to my living room when I see a vocalist but hear only a relatively inferior public address system.

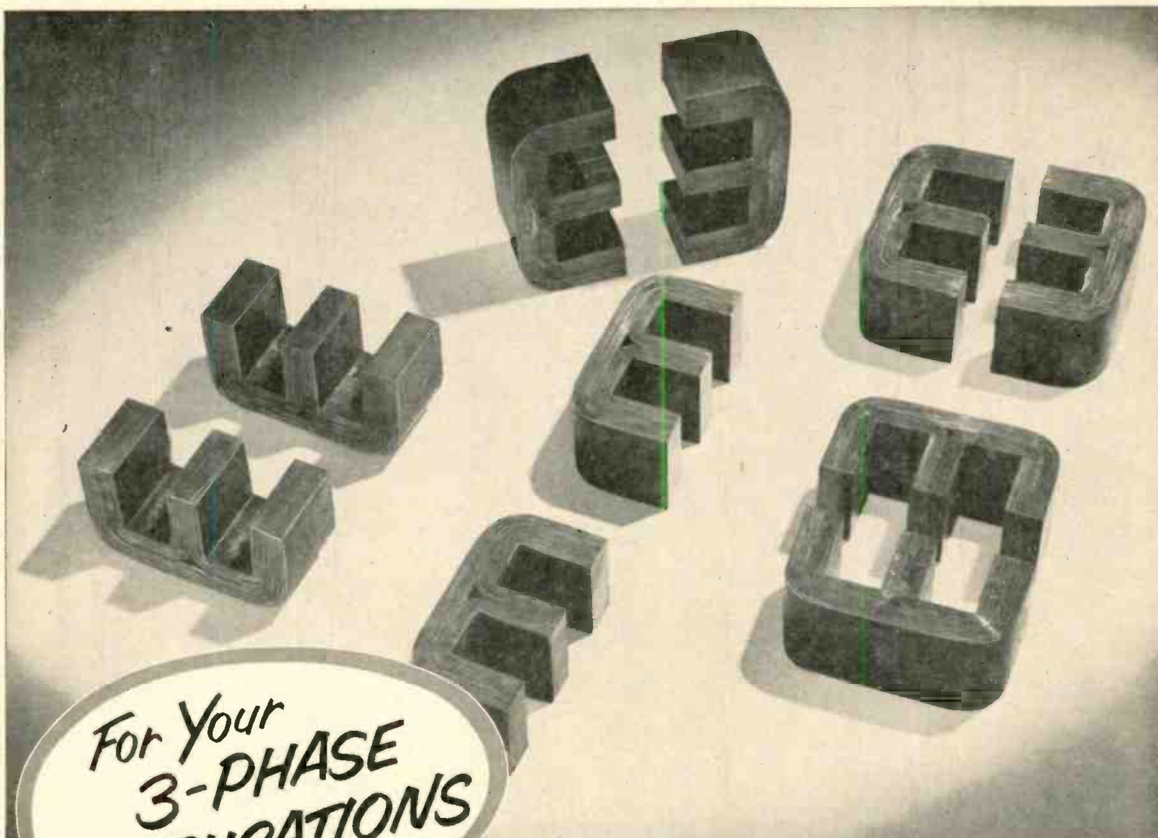
THOMAS O. DIXON
Head, Instrumentation Division
Sound Division, Code 4006
Naval Research Laboratory
Washington 25, D. C.

ORFEO'S NEW MUSIC STUDIO

High-fidelity grown up—might well be the name for the new sound studios opened on November 19th at 19 E. 48th St., New York, for it has introduced a new method of merchandising audio to the customer. Under the direction of Ben Pinz, Orfeo has built two demonstration rooms which are devoid of bare equipment, and which duplicate typical home environments—one modern and the other in traditional decor.



Planned for the Aunt Minnie type of buyer, Orfeo offers a normal living-room arrangement with several "package" systems suitably cabineted, and working as complete units so the non-technical listener may judge performance and can see for himself just how high-quality sound reproduction would fit into his own home. Orfeo's services include complete decoration and acoustical treatment when required, bringing the knowledge of the custom builder and the talents of the decorator together in a unique and workable combination.



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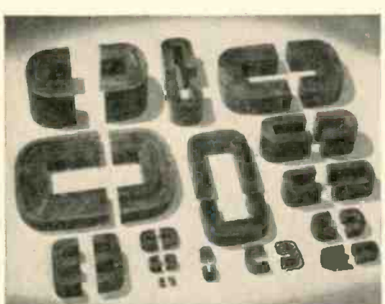

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HAROLD T. SHERMAN*

History and prospects of a new step in the search for optimum realism in sound reproduction.

As time marches on, new developments in audio play an important part in relaying by radio greater realism through a feeling of presence. One of the earliest attempts to transmit this feeling of presence was by means of a dual telephone circuit, using one circuit for each ear, between the Paris Opera House and the Paris Electrical Exhibition in 1881. Binaural broadcasting has demonstrated that the listening audience outside the studio can enjoy the program with less listening effort and less aural fatigue.

A brief history on the subject of binaural broadcasting reveals that the interest of engineers and experimenters was sparked off in the very early 1920's when radio broadcasting was just beginning. At that time the limitations of the audio components used for converting speech and music in the studio into realistic sound in the home created the need for radical changes in sound reproduction methods. In those days the parlor stereoscope with pictures produced by a stereo (binocular) camera were well known for their ability to give us third-dimensional visual reproduction.

Radio fans around New Haven, Connecticut, were probably the first to hear the results of binaural broadcasting. F. M. Doolittle, Professor of Electrical Engineering at Yale University, pioneered this radio system and applied for U. S. patents in 1921. He had articles published in *Electrical World*, and in the radio section of the *New York Herald Tribune* in 1925, giving general information on his findings, with a description and photographs of Station WPAJ, which operated binaurally during 1925. WPAJ used two 50-watt AM transmitters, one on 1320 kc and the other on 1110 kc. One antenna was excited by both transmitters whose antenna coils were connected in parallel.

During this same period, KDKA in Pittsburgh also used Mr. Doolittle's system experimentally and eventually bought his patents. Unfortunately the crowding of the AM band of frequencies made it impractical to continue, or expand, binaural radio at that early period.

In the early 1930's, Bell Telephone Laboratories gave several technical demonstrations of their research on stereophonic sound reproduction for motion pictures. These demonstrations, given to the AIEE, the Acoustical Society of America, and other engineering groups, made a deep impression of the listening advantages, but a practicable two-channel, non-interfering, radio-audio circuit was not readily available.

Present Interest Growing

Revival of development of binaural sound reproduction came when Magnecord, Inc. of Chicago, produced a special twin-track magnetic tape recorder for the Navy Department. It was demonstrated in 1951 that this unit basically was suited to record and reproduce music and sound in third dimension for special music study.

* *Sherman Studio, Carnegie Hall, New York.*

At the Audio Fair in Chicago in May, 1952, Magnecord, Radio Craftsmen, and Jensen cooperated with WGN (AM) and WGNB (FM) to demonstrate that binaural radio broadcasting—using the AM frequency for one channel and the FM frequency for the other channel—was a practical improvement as compared to the single or monaural system. During this binaural broadcast to several hundred hi-fi fans in the hotel ballroom, WGN received several hundred phone calls from home listeners, who for the first time had their ears effectively placed where the microphones were located in the studio. The next day WGN received scores of enthusiastic letters about the improvement in realism, as well as the apparent freedom from distortion of loudness differences. This latter effect improved definition and made it possible for each voice of the orchestra to be followed just as one can do when his two ears are in the microphone location.

Mr. Sherman

It was also discovered that a space consciousness was added to the reception, giving better judgement of the size of the studio and the number of musicians. Possibly the biggest improvement was noticed in the added directional consciousness which spread the orchestra out in the listening area at home, so that you felt the violins in their relative location to the woodwinds, brasses, drums, etc.

Orchestra conductors, choral directors, and organists find that they hear definition and tone-value balance from a listening position more normal to their audience when they review their work binaurally. Orchestras and choirs can be rehearsed by assistants or students, with a delayed reproduction creating a most accurate sound picture for critical review.

AM Used With FM

FM radio offers the logical medium for the second channel in a binaural system because of a complete non-interference with any AM transmitter frequency. Static-free reception and even better audio-frequency

[*Continued on page 71*]

MAURER
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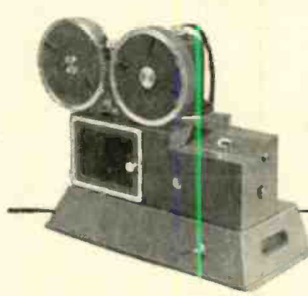

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EDITOR'S REPORT

NEW YEAR'S RESOLUTIONS

Comes the first of January and many of us have the urge to make resolutions. Like many a reader, we felt these urges, and even thought out the first few paragraphs before putting them on paper. Since they seemed somewhat familiar, we checked back to January, 1952, to make sure we were not simply repeating ourselves. Believe it or not, we found many of the same words and expressions, not to mention ideas. If Æ were like some magazines, that would not have been disastrous, since few readers would have had the previous January's copy to compare. But we know that a large percentage of Æ's readers keep complete files, and (assuming anybody reads this page at all) they would probably remind us of the repetition. We like to think that 1952's first Report was well written, and that it contained the credo we intend always to follow. But we must admit that we have not done everything we hoped to during the last year.

One of the changes we aimed for was to attempt to enlighten the novice in our field in the technical aspects of audio. We have made some progress in that direction, but we have not done all we would have liked to. We have included much material that scarcely could be called "engineering" in an attempt to provide the maximum coverage of the entire audio field. As a result, the more technical-minded of our readers were inclined to think that there was too much "butter on the popcorn." Keeping a desirable balance between the technical and the non-technical articles is like walking a tightrope. We *try* to please every one of our readers, and we realize that our future depends on how well we succeed.

On the credit side, we note that in 1952 we had a total of 762 pages—both editorial and advertising—as compared to 640 in 1951, an increase of 19 per cent. This corresponds quite closely to two extra issues, since the average was 63.5 pages. For this gain, we are indebted to our advertisers, for only through their support is it possible to produce a magazine. They are responsible for the 96 and 92 page issues of October and November, and we are pleased that they have seen fit to use more of Æ's pages to carry their messages to you.

And so it is, on the threshold of a new year, that we salute our advertisers. They work the year around to make their products as good as they know how, for they too have to please their customers. Without advertisers we wouldn't exist—and many of our readers would not be so well informed about the products that make their spare time interesting—the products that bring the best of sound reproduction into the homes of discerning music lovers. No matter how the program arrives in our living rooms—by AM or FM radio, by wired music systems, by light waves, or over the water pipes even—it is the program that is important. In the final count, audio equipment must be involved from the moment the original sound is created in the studio or auditorium until it is recreated in our homes. Until someone comes along with a new method of sound reproduction, audio appears to offer the best possibility.

BINAURAL OR STEREOPHONIC?

It isn't often that Æ devotes an entire issue to any one subject, but if any single issue can rightly be called a special one, this is it. From the reception it had at the recent Audio Fair in New York, it would appear that the newest thing in the audio field will involve two completely separate channels—which means twice as many amplifiers, twice as many speakers, twice as many signal sources. In the studio it will involve twice as many microphones in addition to amplifiers, attenuators, loudspeakers, and all the other elements in a transmission system.

Binaural broadcasting is certain to require new techniques—one of the most important being that of microphone placement, for with the present plans which involve both AM and FM radio channels, each program must sound well by itself, since many listeners will not be equipped to reproduce both at the same time. We have not got that far in our coverage of the binaural picture yet, but it is sure to come in the next year. In any case, it is sure to give us an opportunity of comparing systems quite easily—something which many of us don't do with a single set of equipment. This swing toward binaural or stereophonic broadcasting should bring out many new developments in the next twelve months. For the present, however, we hope we have covered the field reasonably well, as evidenced by the articles on pages 14, 22, 25, and 46. We will keep informed on this subject, and will pass on all the information we receive.

SOUND REPRODUCTION COURSE

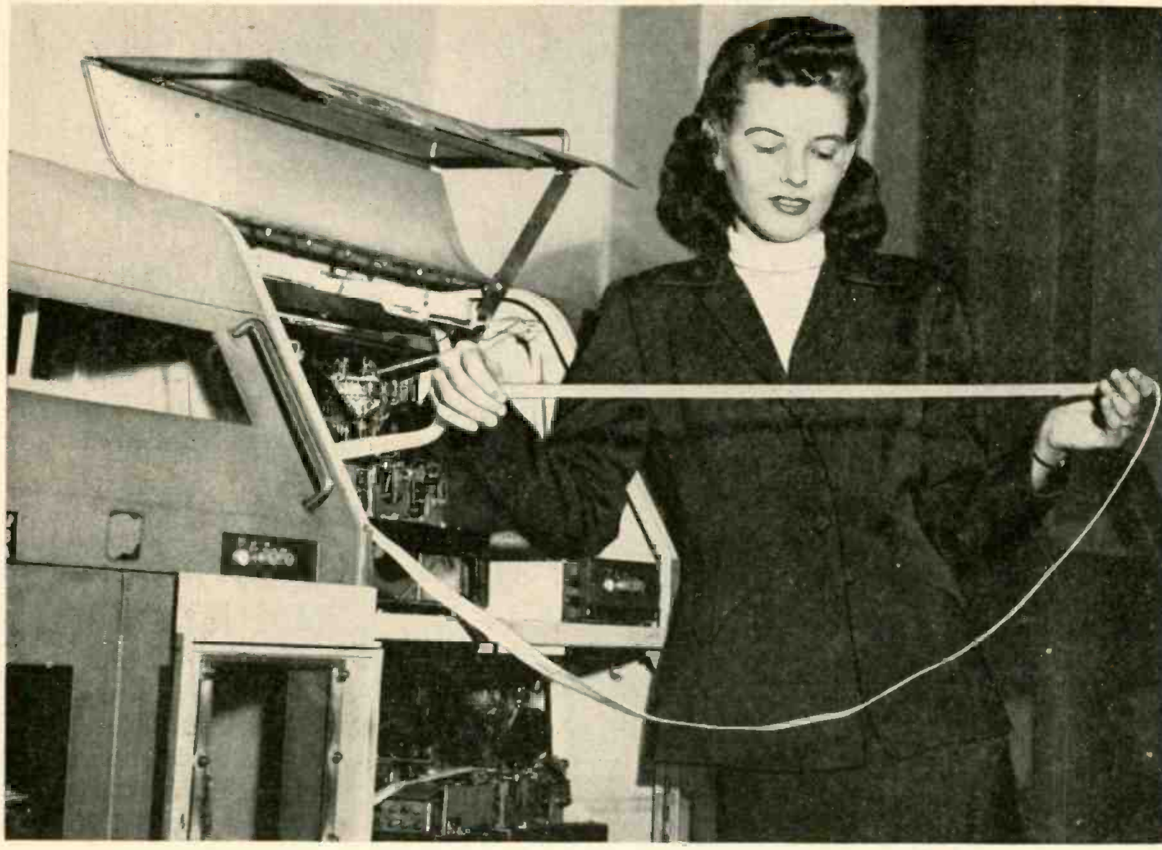
Edgar M. Villchur, who is currently authoring the series entitled "Handbook of Sound Reproduction" is beginning a new course on the subject of sound reproduction at New York University, Division of General Education. This course begins on February 4 and runs until May 20, meeting at the Washington Square center on Wednesday evenings from 7:00 to 9:45 p.m. Further information can be obtained from NYU, and registration by mail or in person begins on January 19.

TAPE RECORDING IN ICELAND

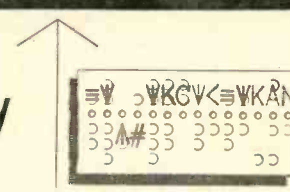
We note with interest that the proceedings of the Iceland Parliament at Reykjavik are to be preserved for future generations by means of tape recording equipment recently installed in Iceland's two chambers of parliament. With eleven Stancil-Hoffman recording units, the entire proceedings of the General Assembly's continuous debate will be recorded. We hope this practice will spread to the Western Hemisphere before many more months.

STANDARDIZATION FOR LOUDSPEAKERS

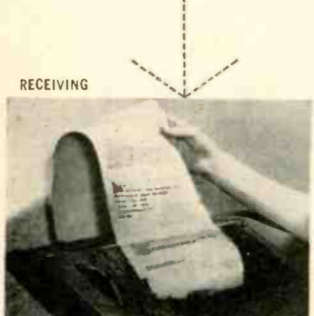
It has been suggested that loudspeakers be standardized as to polarity, and be marked so that two speakers can be connected in parallel without trying to cancel each other. The proposal from one of Æ's readers is that a polarizing symbol be applied to that terminal which, when fed with a positive voltage, will result in an outward cone movement. This seems simple enough, and would certainly help with horn-type tweeters, where it is nigh onto impossible to determine which way the diaphragm moves when we apply a small voltage from a battery.



These signals find the way



SENDING



RECEIVING

When you dial a telephone number, high-speed switching mechanisms select your party and connect you. Through a new development of Bell Telephone Laboratories, similar mechanisms are doing the same kind of job in private wire teletypewriter systems which America's great businesses lease from the telephone company.

Company X, for example, operates an air transportation business with scores of offices all over the country. At one of these offices, a teletypewriter operator wishes to send a message, let us say, to Kansas City. Ahead of the message, she types the code letters "KC". The letters become electric signals which guide the message to its destination.

Any or all stations in a network, or any combination of stations, can be selected. Switching centers may handle 50 or more messages a minute . . . some users send 30,000 messages a day. Delivery time is a few minutes.

Defense manufacturers, automobile makers, airlines and many other American businesses are benefiting by the speed and accuracy of the new equipment – another example of how techniques developed by the Laboratories for telephone use contribute to other Bell System services as well.

BELL TELEPHONE LABORATORIES

Improving telephone service for America provides careers for creative men in scientific and technical fields.

Practical Aspects of the R-J Speaker Enclosure

WILLIAM JOSEPH* and FRANK ROBBINS**

A further discussion of the enclosure which has created so much interest and which was the forerunner of the current flurry for small cabinets.

COMMERCIAL HIGH-FIDELITY reproducing equipment has for some time been available at prices low enough to permit widespread use in home installations. Tuners, amplifiers, cartridges, and speakers which can handle the range from 50 to 10,000 cps are today quite reasonably priced, and equipment handling 30 to 15,000 cps is available in the higher-price range. Speaker development has reached the stage where the former range can now be handled by single-cone speakers and the latter by two- or three-way speaker systems. Most of the progress taking place in speaker development over the last decade has been mainly in extension of the treble range, since low bass reproduction in speaker techniques preceded the development in the treble range.

Reproduction of the low bass end, however, is a function not only of the speaker but of speaker enclosure as well. As almost all 12- and 15-in. speakers designed for the high-fidelity market are capable of reproducing down to 50 cps and below, the responsibility for low bass reproduction today rests almost entirely on the enclosure. Unfortunately, space limitations in the modern home mitigate against the use of large enclosures and in most home systems the octave below 100 cps is generally missing or is strongly curtailed. While such systems would sound balanced and pleasing before the full treble range became available—and do indeed sound better if the amplifier treble control is cut back—it is generally found today that home systems are operated with the treble control flat and with the bass control considerably boosted in an attempt to improve balance. While this boost in the 100- to 200-cps range may help matters somewhat, there is actually no true bass "feeling" to such reproduction. It was considered worthwhile, therefore, to attempt to develop a small enclosure which would permit extending the bass range to at least 50 cps, and if possible to attempt to attain this result without resonant peaks, yet with good damping and transient response. Resonant peaks in the bass range produce "boom" and "barrelly" speech reproduction, while poor transient response "muddies up" the bass.

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A brief consideration of existing systems required to provide 50-cps response will be pertinent in tracing the development of the R-J system and the following types of speaker housings will be considered:

1. Flat Baffle
2. Open-back Box
3. Closed Box
4. Horns
5. Bass Reflex
6. The R-J Enclosure

The Flat Baffle

To reproduce 50 cps adequately, a flat baffle must be eleven feet square. If the speaker itself resonates at 50 cps, there will be a hump in pressure-response curve at this frequency and the response will fall off below this point at the rate of 18 db per octave.

The Open-Back Box

Open-back boxes are not suitable. A 7-cu. ft. open-back box 4 in. from the wall produces a walloping "boom" at 100 cps[1] and response drops off below at the rate of 18 db per octave. Below 100 cps, frequency-doubling and tripling is extremely objectionable.

Closed Boxes and Horns

Completely enclosed boxes need to be very large. A 15-in. speaker will require 12 to 18 cu. ft. of air, depending upon speaker compliance.

Horns provide high efficiency and good transient response. For 50 cps, the horn works out to a mouth diameter of 6 ft. 8 in., or its area equivalent, and a length of about 8 ft. This is a little bulky for use in the average home. However, Klipsch[2] has been able to fold the horn ingeniously so as to require only 13 cu. ft. of space.

The Bass Reflex

In pursuing small enclosure design, it was felt that the most fruitful field for investigation lay in the direction of resonating systems. A resonating column[3] or chamber will produce sound with very little energy to activate it. Consequently, proper design may achieve useful sound output at some point in the register where system response would otherwise fall off. The bass-reflex enclosure[4] is an example of this kind of system which, until recently, offered the closest approach to meeting the space problem. The typical bass-reflex enclosure shown in *Fig.* 1 consists of a box with a speaker mounted in it and a port hole in the front near the speaker opening. The first

[1] Daniel Plach and Philip Williams, "Loudspeaker enclosures." AUDIO ENGINEERING, July 1951.

[2] Paul W. Klipsch, "Design of compact two-horn loudspeaker." *Electronics*, Feb. 1946.

[3] Benjamin J. Olney, "Acoustical labyrinth." *Electronics*, Apr. 1931.

[4] A. L. Thuras, "Sound translating device." U. S. Patent 1869178.

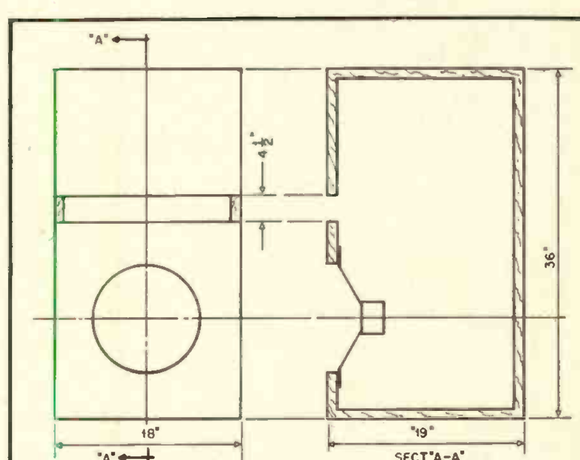

Fig. 1. Typical bass-reflex cabinet, and the one used in the comparison measurements described.

step in the design of the enclosure requires that the air resonance of the box occurs at the free-air cone resonance of the speaker.

The air resonance of the bass-reflex enclosure can be closely determined:

$$f = \frac{c\,(A)^{1/4}}{2\pi\,(V)^{1/2}} \qquad (1)$$

where A = the area of the port
V = the volume of the enclosure
c = the speed of sound in air

This bass reflex cabinet, later used as a comparison enclosure, employs a 12-in. speaker with a free-air cone resonance of 63 cps. To determine the volume V required, the formula may be rearranged to read

$$V = 2070 \left[\frac{(A)^{1/4}}{f}\right]^{1/2} \qquad (2)$$

where V is in cu. in.
A is in sq. in.
f is in cycles per second.

Using A of 75 sq. in. and f of 63 cps, V comes out to be 9400 cu. in. Adding another 400 cu. in. for the speaker, the total internal volume required is 9800 cu. in. or 5.7 cu. ft.

Bass reflex enclosures must be carefully tuned after mounting the speaker to produce optimum results. Since the enclosure is generally purchased separately and the speaker mounted within it without further adjustments, it is seldom that proper results are obtained. This is one of the major drawbacks of the bass reflex. When properly tuned, a bass-reflex enclosure will exhibit two impedance peaks of equal amplitude, equally spaced in frequency above and below the speaker free-air cone resonance. The impedance curve of the comparison enclosure with a 12-in. 8-ohm speaker mounted in it is shown in *Fig. 2*.

Exploring the possibilities for appreciable reduction in size of the bass-reflex enclosure, it becomes evident from inspection of equations (*1*) and (*2*) that this may only be accomplished by reduction of the port area. Basically, the enclosure air resonance must occur at the free-air cone resonance of the speaker. The enclosure air resonance, being out of phase with the speaker, restricts the tendency for large movements of the speaker cone while at the same time sound radiation from the port area takes over and helps to provide acoustic output at this frequency.

If the port area is reduced, acoustic radiation—which is proportional to the port area—drops off correspondingly and the desired sound output is not obtained. In addition, if the port area becomes too small the enclosure begins to approach closed-box performance. There is, therefore, a law of diminishing returns in operation and as a result, ports are generally compromised to about 75 per cent of the speaker area. Sometimes the further addition of a duct added to the port is employed. By this means a further small reduction in volume can be obtained.

The R-J Enclosure

In pursuing the design of a small enclosure, various types of resonant principles were explored and an adaptation of the Helmholtz resonator was finally adopted. In this type of resonator, *Fig. 3*, the mass of air in the opening A swings back and forth at resonance and the air in the cavity V acts as a resisting spring against the movement of the air in the opening. The combination sets up a system analogous to the weight on a spring and has a resonant frequency determined by the mass and the compliance of the arrangement. The equation for the resonant frequency of the resonator is the same as that for the bass-reflex enclosure, equation (*1*).

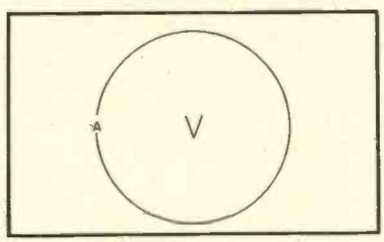

Fig. 3. Basic diagram of Helmholtz resonator.

Helmholtz resonators are characterized by extremely high Q and if a speaker is placed within such a resonator the response curve will exhibit a very high peak at resonance. If, however, some means to control the Q can be incorporated into the system, a small enclosure could conceivably be designed which would permit flattening the peak. A practical form of enclosure which attains this end takes the form shown in *Fig. 4*.

The system still acts as a Helmholtz resonator with the further modification imposed by the creation of a duct system, indicated by arrows in the figure, between the back cavity and the frontal opening. The resonant frequency of an enclosure with a volume of 4800 cu. in. and a frontal opening of 64 sq. in. calculates out as follows:

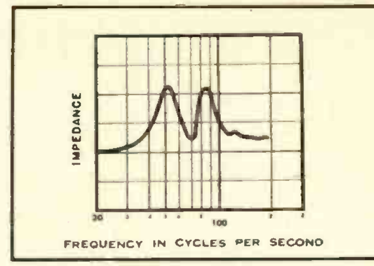

Fig. 2. Impedance curve of the speaker mounted in the cabinet of Fig. 1.

$$f = \frac{(1080 \times 12) \times (64)^{1/4}}{2\pi\,(4800)^{1/2}} = 84 \text{ cps.}$$

Actually, close calculations require several refinements to take into account the effect of duct length and the end corrections.

Control of the circuit Q is obtained by varying the spacing of the duct produced between the frontal board and the speaker board. By decreasing this spacing it is possible to introduce acoustic resistance to lower the Q and increase damping. The acoustical resistance of a slot is expressed by the equation:

$$R_a = \frac{kl}{t^3 w}$$

where k = a constant
l = the length of the duct passage
w = the width of the duct passage
t = the thickness of the duct passage (the spacing between the frontal board and the speaker board.

The acoustic resistance is a function of the third power of the spacing and becomes quite critical as t becomes small. Halving the spacing increases the acoustic resistance by eight times. By experimenting with the spacing it was found possible to reduce the circuit Q and broaden the resonance so that with a speaker of 63 cps free-air cone resonance, this system extends smoothly to 50 cps without any peak at resonance.

The introduction of the acoustic resistance also has a strong effect upon speaker loading. This is very desirable in improving speaker damping and the magnitude of damping obtained may be judged from the amount of lowering in frequency of speaker resonance below the free-air cone resonance. As shown in the impedance curve of *Fig. 5*, the speaker resonance in the R-J enclosure is 32 cps, as against 50 cps for the same speaker in the bass-reflex enclosure, *Fig. 2*. It is believed that a 32-cps loaded speaker resonance, as against a 63-cps free-air cone resonance, is considerably lower than can be achieved by other systems.

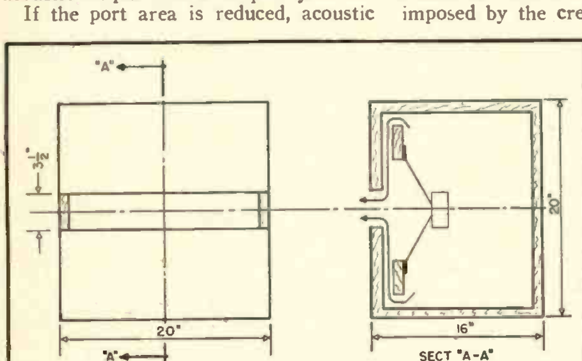

Fig. 4. Front and sectional views of R-J speaker enclosure.

[*Continued on page 67*]

Zero-Impedance Output Stage

RAYMOND G. ANTHES*

Excellent transient and low-frequency response and good loudspeaker damping make this amplifier suitable for high-quality, low-power applications.

THE ZERO-IMPEDANCE STAGE to be described was designed for home use, along with its driver, to give good quality performance at moderate cost. A series R-C circuit (R_1, C_1 in *Fig.* 1) shunts the primary of the output transformer so that the output tube works into almost unity power factor load. This minimizes harmonic distortion and phase shift. The feedback circuits are direct coupled and the negative voltage feedback is taken from the primary of the output transformer rather than the secondary in order to reduce undesired phase shift to a minimum in this feedback loop.

The low-frequency response is exceptionally good because the stage is effectively acting as a zero-impedance source feeding the primary of the output transformer. The output transformer used was of good quality and had 1-inch stack. A frequency response taken with the loudspeaker connected, and measuring output voltage across the secondary of the output transformer indicated the 3-db-down point was below 20 cps at the low end, and at 5000 cps at the high end, and only 9 db down at 15,000 cycles per second. At 2½ watts output into a resistance load at 400 cps, the total r.m.s. distortion was under 5 per cent. This is relatively high by most standards, but quite low for a 6V6.

A disadvantage of taking the negative voltage feedback from the primary of the output transformer is that this feedback cannot correct for the fall-off in high-frequency response in the transformer. The writer prefers to sacrifice some high-frequency response for minimum phase shift in the negative feedback circuit. This assures that the feedback works most effectively, reducing intermodulation distortion to a minimum, giving maximum reduction of harmonic distortion and maintaining a low-impedance source feeding the output transformer, over and beyond the complete audio frequency spectrum. It is possible to compensate for this loss in highs by a fixed equalizer in the preamplifier, but this was not done because the high-frequency loss was not serious. Most preamplifiers incorporate some form of tone control circuit with treble boost which can be used for this equalization.

* *Professor of Electrical Engineering, University of Manitoba, Winnipeg, Man.*

The use of the series R-C network across the primary of the output transformer to provide virtually unity power factor load to the tube is not new. The theory of this is well known. If a series *R-C* circuit and a series *R-L* circuit are connected in parallel as shown in *Fig.* 2, where the *R*'s are equal, it can be proved that the impedance of the parallel combination will be a pure resistance equal to R at all frequencies, if $R = \sqrt{L/C}$. At frequencies above the resonant frequency of the loudspeaker, the impedance measured across the primary of the output transformer with the loudspeaker load on the secondary may be roughly approximated by a series *R-L* circuit. Consequently, within this frequency range, which extends from approximately 125 cps to the highest audio frequencies, the composite load impedance presented to the tube is very nearly pure resistance with small variation in magnitude with frequency.

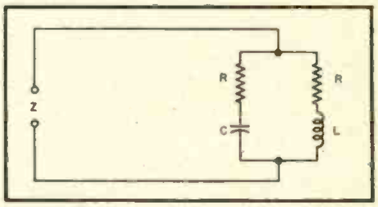

Fig. 2. Series R-C circuit and series R-L circuit in parallel.

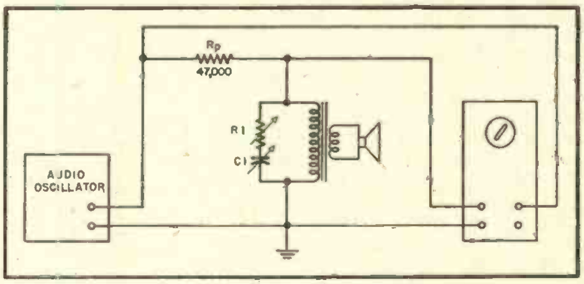

Fig. 3. Circuit used to determine optimum values of R_1 and C_1 of Fig. 1.

If values of R_1 and C_1 are chosen to give the optimum composite load impedance, there will be appreciable reduction in available output power at the higher frequencies where the reactance of C_1 becomes small in comparison with the magnitude of R_1. This is a serious disadvantage. A compromise between these two factors was made in this design.

Adjustment of R and C

The effect of changing R_1 and C_1 can be observed readily on an oscilloscope by the simple circuit of *Fig.* 3, and the values of R_1 and C_1 were finally selected in this way. The value of R_p used was 47,000 ohms, which approximates the plate resistance of the 6V6. The phase angle of the combination is determined from the ellipse appearing on the screen.

For the tube operating voltages used, the load impedance Z presented to the tube should be from 7000 to 10,000 ohms.

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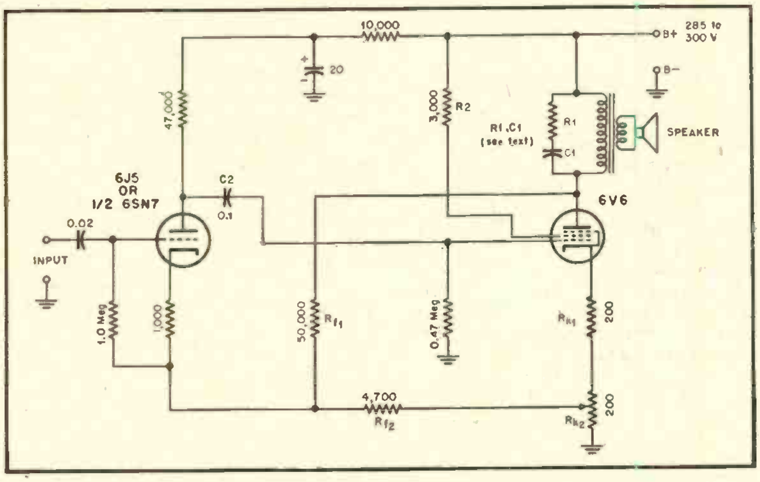


Fig. 1. Schematic of simple two-stage amplifier employing negative voltage feedback and positive current feedback to obtain an output impedance approximating zero.

Binaural or Stereophonic?

R. J. TINKHAM*

The author clarifies the meanings of the two words which have been used to describe this latest of interesting innovations in the field of sound reproduction.

Some confusion apparently still exists in the minds of many regarding the definition and connotation of the words *binaural* and *stereophonic*. Binaural means, literally, "two-eared." It refers to the fact that the sense of hearing in both ears, plus the brain, makes it possible for us to analyze sound, to discriminate against unwanted sounds and, to a large extent, to localize the source of a sound. Stereophonic, on the other hand, means "three-dimensional sound," or a sound source "in the round," much as we perceive it in actual experience. It can be stated, then, that one uses his binaural sense to perceive stereophonic impressions. This is much like saying that one uses his binocular sense of vision to achieve stereoscopic or three-dimensional impressions. If we stop up one ear with a finger when listening to someone in a noisy place, we will observe how the ability to discriminate against the background noise disappears and the ability to localize the source of sound vanishes. Or, if we shut one eye, our judgment of depth is gone and, although we know better, all objects are as on a flat surface. Compare a monocular snapshot with the modern stereophoto or, for that matter, with grandma's "Trip to Egypt" via the old-fashioned stereoscope. The difference is remarkable.

Modern tape recording has achieved, (a) an easy recording method, (b) the ability to record more than one channel at a time on the same tape, and (c) reasonable expense. It is inevitable that someone would think of trying to capture sound stereophonically. Our attention has again been directed to the Bell Labs experiments[1] of 1933 with the Philadelphia Orchestra performing at home and being reproduced in Washington with startling realism, facts lost sight of for nearly a generation. In 1948, Camras[2] gave a demonstration of three-channel recording for Armour magnetic recorder licensees. And in 1950 the author, then with another company, was asked rather bluntly at a Society of Automotive Engineers round-table discussion, why the recorders that he had sold some of the members did not reproduce properly, with the question, "Why does the recording of an auto going over a cobblestone pavement sound like a male singer with a cracked voice humming in the bathroom instead of an auto on cobblestones?" Money had been invested in supposedly good equipment, but the reproduced sounds were not considered natural or normal. Some form of distortion seemed to be present.

If one reflects on the difficulties encountered in achieving "realistic" sound effects for ordinary broadcasting (monaural), where crumpled Cellophane sounds like a raging fire in the north woods and a cantaloupe dropped on the floor portrays the victim's head being bashed in, he realizes that all single-channel reproduction gives only an approximation of the original sound—our imagination fills in the deficiencies. For instance, a symphony over the network sounds like a symphony, with which we are reasonably familiar. But recordings of autos on cobblestones do not sound normal, for this sound is a bit unusual and our imagination does not help out much.

Let us consider the "auto-on-cobblestone" problem briefly. Since the one recording microphone, placed in the driver's position, had no analytical or discriminatory power of its own, it recorded faithfully what it heard and, for the first time (perhaps) the auto researchers heard, in truth, what they didn't hear—a hard fact to realize. Obviously, the solution was to restore the natural sense of binaural hearing by using two complete and separate sound transmission channels, from microphones to headphones, one channel to each ear. This was suggested, and one organization tried it with successful results.[3] Oddly enough, loudspeaker reproduction, at that time, was not successful for reasons which will become clear as we continue.

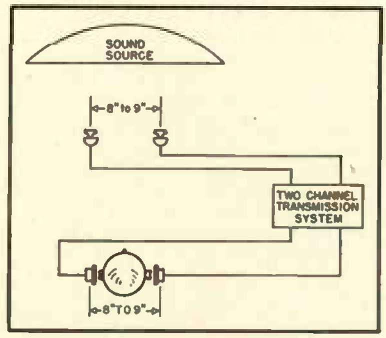


Fig. 1. The elements of a true binaural system.

Microphone Placement

Apparatus intended for one field often ends up in another. Gene Carrington, educational director and lecturer for a large midwestern organization, learned of the foregoing and made orchestral recordings at Interlochen (Michigan) Summer Music Camp under the direction of Dr. Joseph Maddy. Mr. Carrington conducted experiments with respect to the placement of the two mikes with separations varying from many feet to about the distance of a human's two ears. Headphone listening showed that the sound was most realistic when the mikes were about 8 or 9 inches apart. And the demonstration was astonishing in its realism; one seemed to be standing right in the midst of the orchestra. During the past year, several demonstrations of "binaural" sound have been presented, using two loudspeakers—one for each channel—in a large room. The results were puzzling to many people who attended these demonstrations—until they heard similar demonstrations with headphones. Then they understood what was being attempted. Suddenly "three-dimensional sound," whether reproduced binaurally over headphones or stereophonically over loudspeakers, has become most popular. But this popularity has raised another question. Why don't these demonstrations work so well with loudspeakers? After all, it is mighty inconvenient to listen for any length of time to headphones, and loudspeakers reach larger audiences less expensively.

The answer is that loudspeakers do work very well when properly handled. And here entered the confusion between "binaural" and "stereophonic."

If we refer to *Fig.* 1, where we have two closely spaced microphones in a sound field, and a two-channel system connecting them to separate earphones. it becomes obvious that we are merely extending the diaphragms of our two ears electrically. Actually, of course, we are extending the diaphragms of the individual headphone receivers, electrically, to the physical placement of the diaphragms of the microphones. The time factor may be suitably delayed by the insertion of a simultaneous, two-track recorder in the bilateral transmission system. In other words, our ears seem to be where the microphones are physically placed. We have re-oriented our ears geographically and individually. The theorem of proportionality exists: mike-to-mike spacing is equal to the earphone-to-earphone spacing, and our brain functions normally to fuse the two

* *Ampex Electric Corporation, Chicago Office, 111 E. Ontario St., Chicago 11, Ill.*

[1] *Electrical Engineering*, Jan. 1934, Vol. 53, No. 1, six articles.

[2] Marvin Camras, in *Proc. I.R.E.*, April 1949, Vol. 37, No. 4.

[3] *J. Acous. Soc. Am.*, Nov. 1952, Vol. 24, No. 6.

separate signals arriving from each of our two ears. The resultant sensory perception is astonishingly like listening in the spot where the mikes were placed. Of course, we are assuming that we have paid attention to the proper phasing of the mikes, amplifiers, and headphones, and haven't interchanged sides. Disturbing but interesting effects result from such maladjustments. And the quality of the reproduced signal, as always, depends on several factors present in the system: frequency response, distortion, noise, flutter, wow, etc.

But we soon discover that listening to headphones makes the sound appear to be behind, rather than in front of us. This is a fact brought out by Bell Telephone's "Oscar" exhibit at the 1933 Chicago Worlds Fair and now permanently displayed at the Chicago Museum of Science and Industry. And it is also just as tiring today to be squeezed by a headset as when the crystal radio was in vogue. The apparent answer to the problems is obvious. Reproduce the signals over loudspeakers, of course. This will put the sound in front of us, and will be more convenient for listening. But while the sound as we hear it from the two loudspeakers sounds different from single channel listening, it still doesn't sound quite right. Remember, we still have the mikes placed only 8 or 9 inches apart. We have suddenly invented a new system of listening: "bistereonauralphonic" or a cross between two different concepts—which is meaningless. See *Fig.* 2. The placement of the speakers is unimportant here, as no arrangement will give the desired effect. It doesn't matter whether we set the speakers at an angle so that their axes intersect, as at (A) in *Fig.* 2, or so that they are square with each other. Speaker manufacturers pride themselves today on the non-directionality and wide-angle dispersion of their products. What happens under these conditions of close mike spacing and wide speaker spacing? If we stand somewhere in front of the speakers, we no longer receive sound meant for the left ear only in the left ear, and right-ear sound only in the right ear, but rather we receive sound in both ears meant for individual ears only. Our original mike placement simulated our relative ear positions, not the loudspeaker positions. This results in a brand new sensory experience which puzzles that superb analytical instrument called a brain, for the sound is twice mixed: once in the air, because each ear hears sound from each speaker, not just from one as it should; and once in our brain. Since we hear left and right sound in both ears simultaneously, with head diffraction lags, etc., our brains are led to the mental conclusion that we hear two speakers. This leads to the inescapable conclusion that we are befuddled, and this type of reproduction becomes a new and novel experience to be sure. This is certainly not the type of reproduction for which we search, namely that of reproducing the original sound stereophonically (i.e. "in the round"). But with this system there is, however, one point or group of points lying on the perpendicular bisecting vertical plane between the two speakers where, due to phase relationship and standing wave patterns in the reproducing room, we can hear sound from the left speaker predominantly in the left ear and vice versa. Here the results are somewhat more satisfactory. A slight shift of the head will make this interesting phenomenon disappear, and again both sound sources will be heard in both ears simultaneously. This narrow central "binaural" plane limits the number of happy listeners to those who can stand close together in tandem.

Corrective Measures

The answer to this predicament is immediately obvious to anyone familiar with acoustics. In the case of the binaural setup of *Fig.* 1, we are not concerned with listening room acoustics, obviously, and this occasionally has its advantages. Also, we are not too much concerned with the recording studio acoustics, since our built-in analyzer (brain) compensates for almost all of the acoustic shortcomings of that room, such as excessive reverberation, unwanted echo, etc. This we have learned to do unconsiously since earliest childhood We use here our binaural sense in the proper and usual manner.

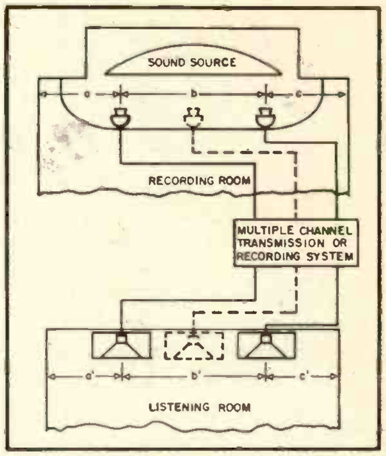


Fig. 3. The true stereophonic system. When three channels are used, distance becomes greater than when only two channels are employed.

But now, since we wish to reproduce this sound over speakers, we must take the listening-room acoustics into consideration. In fact, the listening room becomes an extension of the recording room. Of course, this has always been true in the case of a one-channel system, but not nearly as obviously so as it now is in the case of the two- or three-channel stereophonic system. After all, we are now trying to re-create that special effect in the listening room which existed in the concert hall.

Let us suppose for a moment that we wished to reproduce a concert, by means of loudspeakers, in the same hall as that in which it had been previously recorded. In reality we should have an infinite number of speakers arranged across the stage and several layers high in order to reproduce a similar full wave front. But for the sake of economy we use only two (or three) and spread them, one to the left, one to the right (and preferably another in the center). If we space two of them at, say, the one-third points across the stage as in *Fig.* 3 where $a=b=c$), then, by proportionality, the microphones used to pick up the original sound should have occupied the same positions now occupied by the speakers. Sound is then radiated from the points where it was originally picked up. This is a most important factor, a factor relating to the theorem of proportionality. The recording system has merely delayed the element of time. The speakers then deliver approximately the same acoustic pattern into the room that was picked up at those points by the microphones. Sound from all over the stage reaches the various

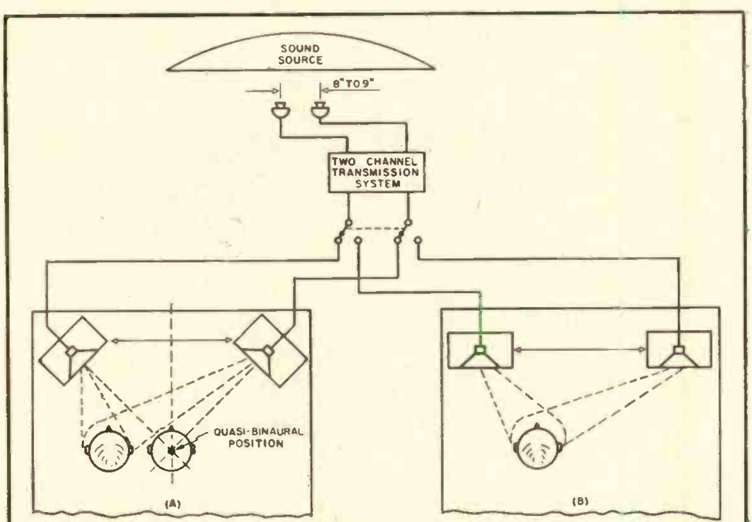


Fig. 2. "Bistereonauralphonic," or mixed and meaningless system.

Fig. 4. A complete portable stereophonic recorder, with amplifiers and power supply.

microphones at varying sound levels and times. These factors are necessary to the proper re-creation of the speaker-reproduced pattern.

If the microphones had been spaced "ear distance" (i.e. about 9 inches apart), as was done in a recent demonstration, and the loudspeakers a similar distance apart to maintain proportionality, rather a neat trick which was not done, we would have had in effect a one-channel system. This would accomplish little toward our ends. But with the speakers spaced widely, we lose the necessary proportionality and achieve only our unrealistic "bistereonauralphonic" system. Conclusion: the mikes should be spaced as far apart for this type of recording (stereophonic) as the speakers are to be spaced. We must know beforehand what sort of reproducing system is to be used.

Next we add the problem of placing the speakers not in the original room where the sound was recorded, but rather in some other and usually much smaller room, perhaps a living room, whose major dimension is less, perhaps than the distance between the recording microphones. Should we, therefore, reduce the spacing of the mikes? Experiment says no. Experiment also shows that if we space the speakers in approximately the same general lateral arrangement in the smaller room as in *Fig.* 3, where $a' = b' = c'$, maintaining approximate proportionality, the results are satisfactory. In fact, the sensitive central plane of "binaural" position, referred to previously, disappears and the auditor is free to move anywhere within the listening room He will experience a similar sensation as though he were to move to various parts of the original auditorium. Placing the speakers in the corners of the room facing diagonally inward or placing them close to side walls appears not to work as well as spacing them a little way in from the walls and square with the room, as in *Fig.* 3. In so doing apparently a more normal sidewall reflection and multiple image pattern is set up in the room, thus simulating more closely the recording set-up. This helps blend the sound pattern in the room. Corner speakers do not set up such an image pattern.

In general, it is necessary to know where and in what manner the sound is to be reproduced before the proper microphone placement can be stated. For headphone listening, mikes should best be spaced a person's head-width apart. A bag of sawdust between the mikes, simulating one's head, is of unknown

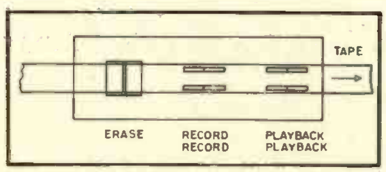


Fig. 6. Diagram of the arrangement of the Ampex stereophonic head assembly.

value. For loudspeaker listening the acoustics of both the recording and listening rooms must be taken into consideration without fail. A wide spacing of the mikes is indicated so that the theorem of proportionality will be maintained. When two channels are used, the mike spacing and position should be chosen with respect to the group size and arrangement so that all sections of the group will appear balanced in the reproduction. An overly wide spacing of the mikes will leave a "dead" spot in the pickup between the mikes (and speakers) at the center of the group. This is where the third channel, in the center, becomes helpful by really rounding out the stereophonic system. The Bell Labs experiments[1] showed that three channels were an optimum minimum number, but that two would work satisfactorily, if properly handled.

The insertion of a dual AM and FM simultaneous broadcast radio link does not change any of the conditions set forth, obviously, but does limit the system to two channels. Separate-sideband AM transmission plus FM has been suggested for a three-channel system, but appears to be still in the future.

It should be repeated that the quality of the total system will limit the naturalness of the reproduction. This was clearly demonstrated at the November, 1952, concert of the University of Illinois Symphony Orchestra, conducted by Leopold Stokowski, at Urbana, Illinois, where the author was invited to transmit the concert stereophonically to an overflow crowd in a relatively large but acoustically different auditorium on another part of the campus. Stereophonic recordings were made simultaneously. During the rehearsal two cardioid mikes were tried, followed by two modern condenser mikes. The former, designed many years ago, have an increasingly attenuated response above 9,000 cps. The condenser mikes have good high end response. Several of the musical faculty present commented on the noticeable difference, and the more nearly true string and oboe tone resulting from the switch. No other elements of the system were changed.

One controversial point might be raised here. What mike pickup pattern should be used? The author prefers a cardioid pattern, but with a wide frequency response. This pattern seems to help stereophonic reproduction because the pickup thus is one-sided, while the speaker reproduction is other-sided. This would seem to help in making the listening room an extension in fact of the recording studio.

[*Continued on page 57*]

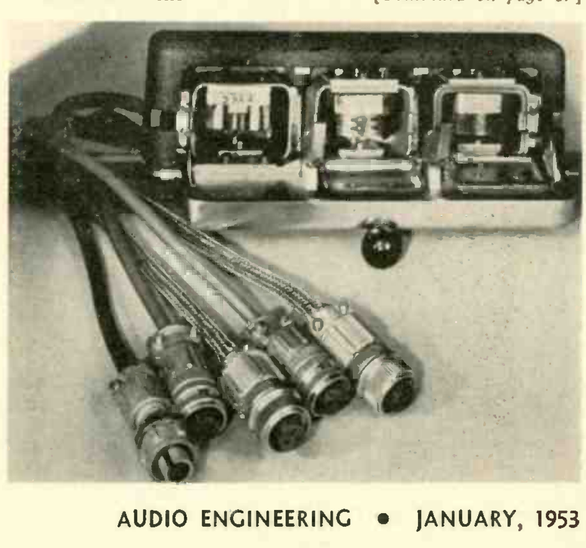


Fig. 5. Stereophonic head assembly, employing a full-track erase head at the left, a two-track record head, and a two-track playback head at the right.

Two-Channel Conversion for AM-FM Receivers

C. G. McPROUD

Details of simple changes which give the advantage of two separate receivers to tuner chassis which employ independent AM and FM channels but with coupled tuning capacitors. Ideal for binaural or stereophonic broadcast programs.

Some time ago, the writer made a few changes to a standard AM-FM tuner chassis in order to be able to listen to one program on FM while recording another on AM, or vice versa, all without realizing how advantageous this modification would be when radio stations started to use their two transmitters to radiate the two-channel signals for binaural or stereophonic programs.[1]

The whole idea is so simple and straightforward that anyone faced with the problem of providing both AM and FM programs simultaneously from a single tuner should come up with the same solution. However, since such a solution has not yet appeared in print, this one is offered as one way to accomplish the desired end.

At the time the modification was planned, there were no stations regularly broadcasting two-channel programs, but it must be admitted that there are times when two programs of interest are on the air at the same time, and the human hearing and interpreting mechanism is unable to cope with both at once. For this problem, the simplest solution is to record one of the programs. As one element of the writer's home system, a wire recorder was employed for just this purpose. The quality left something to be desired, but it was at least good enough to convey the information contained in the program—probably as well as any of several million table-model sets that serve as the sole entertainment medium in many households. At one time, this system employed two separate tuners—originally a wide-range t.r.f. tuner of the Miller type, with modifications of course, and the other an inexpensive FM tuner with a ratio detector.

The next step in the program of improvement included the design and construction of separate AM and FM tuner chassis, both tuned by means of push buttons. The FM tuner used relays to switch adjustable capacitors in the oscillator circuit, while the other used relays to switch the plate supply to any one of four single-channel fixed-tune chassis. This entire system was described in an earlier series of articles.[2]

The next step in the program was to secure a combination AM-FM tuner of good quality, with the idea of separating the sections so as to have the advantages of two-channel reception if desired. Because of its physical construction, the Browning RJ-12B was selected. In this receiver—as in several other types—the tuning is accomplished by two entirely independent variable capacitors, coupled together only by means of a dial cable. The modifications are now reasonably obvious. With the advent of two-channel broadcasting for binaural or stereophonic programs, the receiver becomes doubly useful.

Before discussing the actual modifications to the RJ-12B, let it be said that the general idea of these changes can be adapted to any AM-FM tuner which uses separate devices to tune the two sections. The Browning RJ-20 is similar to the RJ-12, and the Meissner 9-1091 may be converted in a similar manner. The details may differ, but the principle remains the same. For simplicity, only the changes to the RJ-12 will be described, and they are divided into two parts—electrical and mechanical.

Electrical Changes

The electrical changes required are simple, and are made primarily to permit the simultaneous operation of both channels. In the system for which this modification was made, the switching between AM, FM, and Phono is accomplished in the control amplifier. Therefore, since the phono and the TV inputs to the tuner were never used, it was preferred to eliminate this switching facility from the tuner chassis. However, to reduce current drain and consequent heating, a selector switch was used to permit operation of either channel separately or both together. This switch has three positions—the center providing for both AM and FM operation, the left position energizing only the AM tuner, and the right position energizing only the FM tuner.

In addition, neither the volume control nor the power switch were needed on the tuner chassis, both being provided elsewhere in the system. Therefore both of these were removed. As a matter of fact, when the RJ-12 is to be used in a typical modern system which incorporates a control amplifier, both of these components may be removed, leaving only the two central knobs on the tuner chassis—one for tuning and the other as the selector. However, for the modification, two tuning shafts were needed in addition to the selector switch, so in order to preserve symmetry, the a.f.c. switch was wired to the front panel, which is somewhat more convenient than its normal back-apron location. Thus we have four control positions on the front of the chassis—a.f.c. switch, tuning, selector, and blank. The original controls were, in the same order, the power switch, tuning control, selector, and volume. *Figure* 1 shows the

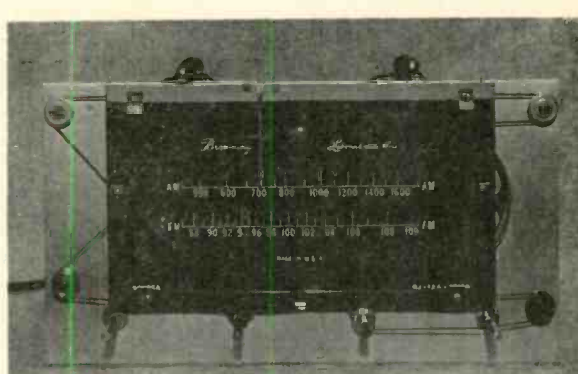


Fig. 1. External appearance of the converter Browning RJ-12B tuner, showing the two dial pointers.

[1] With no intention of getting into this controversy as to the correct name—at least not in a constructional article—the writer respectfully refers the reader to discussions on this subject by Tinkham on page 22 and Canby on page 46.

[2] C. G. McProud, "Elements of residence radio systems," AUDIO ENGINEERING, Sept.-Dec. 1948. Reprinted in AUDIO ANTHOLOGY.

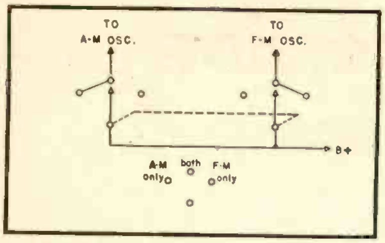


Fig. 2. Schematic of the switch connections to be made in making the conversion.

front of the modified chassis, which is but slightly different from a normal chassis. It will be noted that there are two dial pointers—the one at the top extends only to the AM dial scale, while the new one at the bottom extends upwards to the FM dial scale.

Taking the changes one by one, the following procedure should be followed. Remove the power switch and its associated wiring. Remove the volume control, leaving the wires disconnected but at the same location temporarily. One of the r.f. coils will have to be pushed slightly backward to permit the control to be removed. If done carefully, no damage should result. Now, remove the selector switch, leaving all its wiring intact. This will save a lot of tracing when connecting the new switch. All the shielded wires from the switch to the jacks on the rear apron may be removed, making it easier to get the switch out of the way. Next, remove the tip jacks marked AUDIO OUTPUT and RECORDER INPUT—the first and third from the left end of the rear apron when viewed from the back of the chassis. In the two holes, mount two 1.0-meg audio-taper potentiometers, wiring the arms to the adjacent tip jacks, and the ground end to a chassis ground lead. Connect a .05-μf capacitor from the junction of R_{11} and R_{13} (in the Browning diagram) —from which a shielded lead to the switch was removed—to the high end of the potentiometer nearest the AM end of the chassis. Connect another .05-μf capacitor from the junction of R_{37} and C_{41} to the high end of the other potentiometer. These two potentiometers permit adjustment of the outputs to equal levels.

Install a two-pole three-position switch (Centralab 1462) in the hole from which the selector switch was removed, and connect the new switch as shown in *Fig. 2*. The remainder of the leads should be connected so that the indicator tube works on both sections of the receiver at the same time. To accomplish this, connect the lead from pin 5 of the 6AL7 socket to the AM a-v-c bus; connect pin 4 to the ground end of the discriminator network; and connect pin 6 to the "high" side of the discriminator output, through a 1-meg. resistor. Thus the plate supply is connected to the AM oscillator in the left and center positions of the switch, and to the FM oscillator in the center and right positions.

New A.F.C. Switch

The normal a.f.c. switch in this chassis is a single-pole double-throw slide switch mounted on the rear apron. By prying up the tabs on the sides of this switch, the slider element may be removed and upon replacing the rear bakelite plate of the switch, the wiring is not disturbed and the termnials serve merely as tie points. Install a single-pole double-throw rotary switch on the front apron where the power switch was, and connect three wires from the old switch to the new one, making sure to maintain the same operating arrangement.

After cutting the shafts to the same length as the original tuning shaft, the electrical changes are completed. One hole remains unoccupied on the front apron—that from which the volume control was removed.

Mechanical Changes

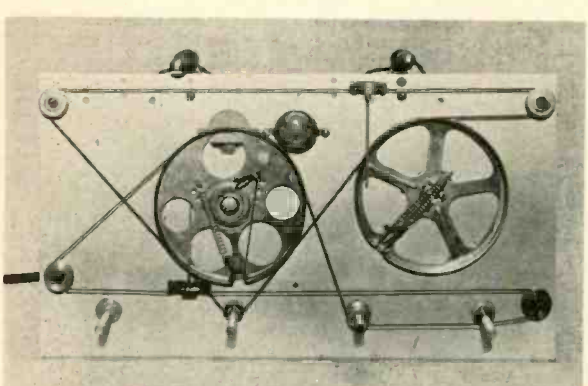
The first step is to remove the dial and all the dial stringing. Then remove the large pulley from the AM tuning capacitor shaft, and remove the smaller pulleys from both capacitor shafts. Mount a 1/4-to-3/8 shaft extension on the FM capacitor shaft, and firmly attach a new 4-in. pulley to this extension, making sure that the dial cable opening in the pulley is at the bottom when the capacitor is half meshed. Remount the 4-in. pulley on the AM capacitor shaft, with the groove of the FM pulley about 1/16-in. further from the panel than the AM pulley. Mount a tuning shaft of the same type as the present one in the hole from which the volume control was removed, and mount two 3/4-in. idler pulleys as shown in *Fig. 3*. These pulleys must be free to turn easily, and eyelets for mounting them are usually supplied with the pulleys. Note that the bottom of the left idler is on a line with the top of the right one, and that this line is slightly above the bottom edge of the glass dial plate. A third idler pulley should be reamed out to run freely on the 1/4-in. selector switch shaft.

Following the diagram of *Fig. 4*, carefully restring the AM dial cord. This cord is shown in solid lines and should be drawn up so as to be quite tight, with the spring extended to about 1½ times its normal length. Where the dial cord wraps around the knob shaft, it is suggested that three turns be taken. Test the stringing before proceeding, making sure that the knob shaft turns freely and that the dial cord behaves properly in the shaft depression.

Following a similar procedure, string the FM cord, using the dotted lines of *Fig. 4* as a guide, and again test the operation thoroughly before proceeding further.

It is suggested that two similar dial pointers be obtained—the writer prefers the type which are made of a fluorescent plastic as they are easily seen. Since one is to slide along the top of the dial plate and the other along the bottom, try them out first, and cut off short enough that they clear, yet long enough to reach the dial calibrations. Then remount the dial plate and attach the top pointer, using a few drops of radio cement in addition to crimping the back of the slider. The vertical line at the left end of the dial scale indicates the position of full meshing of the capacitor plates for both sections. Therefore, the pointer should be secured to the dial cord at this end of the dial and with the capacitors fully meshed.

The FM pointer is similarly mounted to slide along the bottom of the dial plate. This may take some careful adjustment to make sure that the pointer passes the other cords without catching, but it can be done with a little care. It is suggested that the sliders be given a light coating of Lubricate or some similar lubricant.



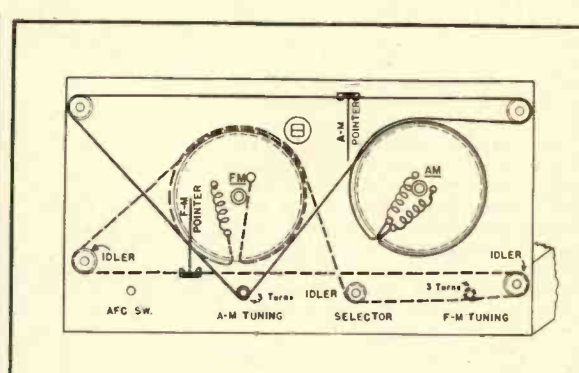


Fig. 3 (left). The tuner chassis with the dial plate removed to show the dial stringing. Fig. 4 (right). diagram of the dial stringing. The solid line indicates the path of the AM cord and the dotted line shows the path of the FM cord.

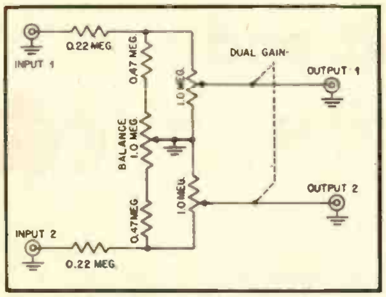


Fig. 5. Schematic of simple control for stereophonic programs.

Output Circuits

For use as a tuner capable of receiving two separate monaural channels, nothing further is required, since the two output jacks on the rear apron provide the AM and FM signals independent of each other. However, for binaural use, some means of controlling the volume of both channels simultaneously is an advantage. Therefore it is recommended that a separate control unit be provided. This unit can be as simple as possible, containing only a dual potentiometer, relying upon the two output controls on the tuner chassis for balancing levels between the two sections of the receiver. If a more elaborate unit is desired, so that either channel can be connected to either power amplifier or to both at will, it is equally possible to make such an arrangement. In any case, if the unit is to be used primarily for stereo programs, the balancing control is desirable. *Figure* 5 shows a simple control, without any provision for switching, yet equipped with the balancing feature. This entire unit can be mounted on a small bracket or on an unused space on the front panel of the installation. To reduce the number of *apparent* controls, it is suggested that the complete unit be assembled using an IRC Concentrikit, with the balancing control operated by the outside knob, and the dual potentiometers operated by the small inside knob. The parts list shows the components for either type of construction.

A more flexible control system is shown in *Fig*. 6. The selector switch has five positions—(1) both AM and FM channels feeding separate output circuits, with AM on output 1 and FM on output 2; (2) same as (1) except that AM is on output 2 and FM on output 1; (3) AM feeding both output circuits; (4) FM feeding both output circuits; and (5) stereophonic, with the dual volume control and the balancing control in the circuit. The only eventuality not provided for is the reversal of the stereophonic sources, which could become necessary in certain instances. This was not included because of the impracticality of locating a four-pole six-position switch of reasonable dimensions. However, it is probable that some standardization in channel usage will occur, so that the FM channel is always on the left and AM on the right, or vice versa. Since the normal orchestra arrangement places the high-frequency instruments predominantly on the left, it is probable that FM would be the left channel.

The balancing control provides a loss of about 3 db in each channel in the center position, with a variation of 3 db in the level of either channel as the control is rotated from one end to the other. If the control unit is to be used with amplifiers at any appreciable distance, it would be desirable to add two cathode followers, as shown in *Fig*. 7. There is some additional gain in this circuit so the balancing control has a mid-position loss of 6 db in each channel, with a range of 6 db from one end of the control to the other. The output from this arrangement is approximately the same for all positions of the switch, and one additional position has been added to permit reversal of the sides in the stereo position. The output impedance is sufficiently low that amplifiers may be located up to 20 feet from the control unit without appreciable frequency discrimination.

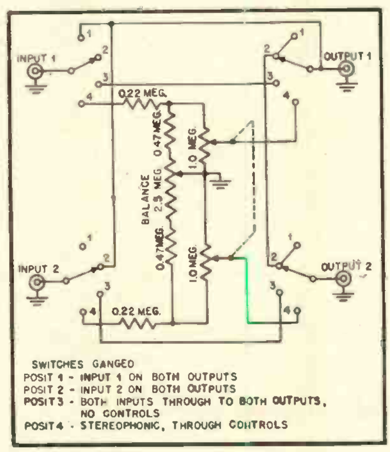


Fig. 6. The addition of a switch provides greater flexibility in the operation of the converted tuner. With the switch at A, input 1 is connected to both outputs; at B, input 2 is connected to both outputs; at C, each input is connected to a separate output, without the volume and balancing controls in the circuit; at D, the inputs are connected through the controls for binaural programs.

PARTS LIST

Receiver Modifications

1 2-pole, 3-position switch, Centralab 1462
1 SPDT rotary switch
2 1.0-meg audio-taper potentiometers, small size
2 .05-µf capacitors, 400 v., paper
1 Dial drive shaft with panel bearing
3 ¾-in. idler pulleys
1 4-in. dial pulley, with tension spring
1 Shaft extension, ¼-in. hole, ⅜-in. shaft
2 Slide-rule dial pointers
Nylon dial cord

For Figure 5

1 1.0–1.0 meg. dual potentiometer, audio taper
1 2.5-meg. potentiometer, linear or, if concentric control is used
1 IRC K-2 Concentrikit
1 IRC KS-2 Universal Shaft Kit
1 IRC B11-239 base element
1 IRC B13-137 base element
1 IRC M13-137 base element
4 Input jacks, RCA Phono type
2 0.22-meg. resistors, ½-watt
2 0.47-meg. resistors, ½-watt

For Figure 6

Same parts as for Fig. 5, with the addition of
1 Centralab 1414 switch

For Figure 7

Same parts as for Fig. 6, except for the substitution of the following for the resistors listed:
2 0.27-meg resistors, ½-watt
2 0.39-meg resistors, ½-watt
4 0.47-meg resistors, ½-watt
2 0.1-meg resistors, 1-watt
2 10,000-ohm resistors, 2-watt
2 3900-ohm resistors, 1-watt
2 Noval sockets
2 12AU7 tubes
2 0.1-µf capacitors, 600 v., paper.

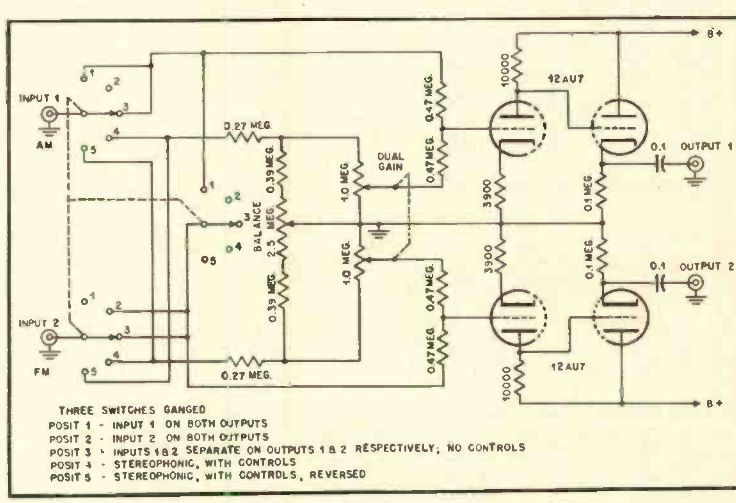


Fig. 7. Similar switching plus the addition of two cathode followers may be preferred because of the lower output impedance offered with this arrangement.

A Flexible Single Recording and Re-Recording Channel

HAL MAGARGLE*

A typical motion picture studio sound system condensed to meet the requirements of a small plant with a need for great flexibility with a minimum of equipment.

THE AUTHOR HAS FOUND in his too short an existence in the business of recording sound motion pictures, especially in the transmission end, that there are books available if you want to delve into the basic theory of recording, and that there are also quite a few allied books available on a high engineering level. However, when the time came to design a recording channel, the needed books were not available to the writer, and he had to take his knowledge piecemeal or by word of mouth as best he could. This paper describes the channel which finally evolved—one which, in the writer's opinion, can be classed as an optimum single recording channel which will perform every duty necessary for the producer with a limited amount of space and equipment inventory. As mentioned, it is a single channel but, of course, can be used to record on different media simultaneously.

The entire channel centers around the RCA MI-10238 Main Recording Amplifier which was modified by RCA to include a jack bay in the lower part of its sub base. This jack bay consists of a single strip of jacks upon which are terminated the compressor, high- and low-pass filters, film equalizer, and trunk lines, as well as miscellaneous other jacks as noted on the schematic. There jacks were installed mainly to facilitate test duties on the unit; however, they make an excellent position from which to evolve the rest of the channel.

Basic Equipment Layout

Following the schematic—*Fig.* 1—from the sound sources to the recorders, we note the microphones and film and

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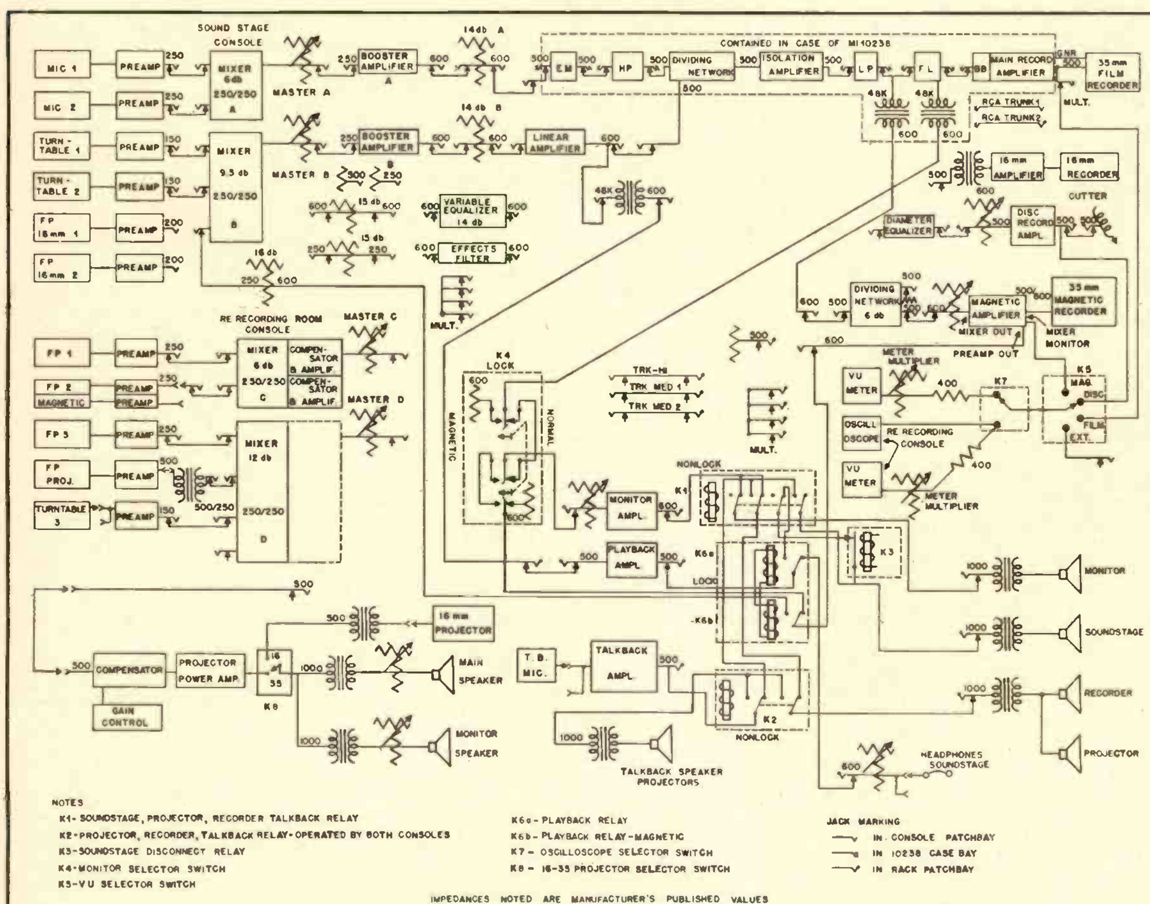

Fig. 1. Overall block schematic of the flexible single-channel sound recording and re-recording system.

disc phonograph machines. These units are conventional, and in all cases the preamplifiers shown are located as close as possible to the microphones or machines, with the signals then being piped into their respective mixing pads. To eliminate a large console, with a pad for every unit, each preamplifier is terminated on jacks and then the most popular set-up is normalled along the line. Because the channel must be versatile and one man—the mixer—must be in complete control of the signals, the entire patching needs are taken care of in one localized spot, again with a minimum of units to operate. In fact, the majority of the jobs done with the channel need no patching whatsoever, and in the remaining cases only a few patches are needed. This is with exception of the re-recording room which has to be patched into the circuit whenever needed. Even here, only two patch cords are needed to change the channel completely from the scoring stage to the re-recording room, everything else being normal. The patches involved are for the mixer positions only.

After mixing in the respective mixing positions, the signal is amplified conventionally and fed to the main recording amplifier. At this point is encountered another modification to the MI-10238 amplifier. The original amplifier could receive one signal only, and either compress it or pass it through in a linear fashion. With the modification, one signal can be compressed, if desired, while the others are sent through linear. The linear amplifier could be replaced with an external electronic mixer (compressor) but in this case it was not needed. All original recording is compressed, with the exception of the location sound which is recorded on the RCA PM 62-A 35-mm magnetic film recorder. This unit has no compressor, and one has never been found necessary because of the latitude and range of the magnetic medium. When the original sound is later transferred to the optical film for the subsequent cutting and dubbing operations, it is then compressed. This is a very satisfactory arrangement because the background noise of magnetic recording is so low that the over-all level in the transfer can be raised by compression almost as much as wanted. Without exception, the original signals on disc or optical film are compressed before actually being recorded. Any of the other recording systems—35-mm magnetic (which is normalled through), disc equipment, or 16-mm recorders—can be patched into the compressor unit of the main recording amplifier. The amount of compression to use is selected by the nature of the sound being recorded, with the compressor set up for limiting duty only during re-recordings. This holds true in almost every case—the exception being "news reel" type re-recordings where the narration is compressed while music and sound effects are left normal. This type of operation is quite simple for this channel and can be set up in the same amount of time as that necessary to set up for voice recording only. When a special type of job is encountered, multiple jacks, attenuator pads, and loading resistors are available for use in these set-ups, if needed, as are two external filter equalizer units—one a variable equalizer to boost or attenuate the highs or lows as needed, and the other a variable sound-effects filter for telephone filtering and similar uses. These two units together are effective in forming a frequency response to match practically any curve needed, including dialog equalization and 16-mm or TV film characteristics. They may also be used to advantage in disc recording work. All the disc equipment is sync, and is direct drive, so that it may be used for any lip-sync or direct sound as needed. The equalizers used for re-recording are the standard-type compensators, and they are used *only* for re-recording. As such, they are an integral part of the re-recording console.

After the signal is combined at the output of the electronic compressor and linear amplifier it passes through the high-pass and low-pass filters and the film equalizers, and thence to the 35-mm recorder. The monitor amplifier is bridged in the circuit ahead of the film-and-slit-loss equalizer so that it needs no compensation and can be used for other duties besides monitoring. During re-recording, the amplifiers in the re-recording theatre are compensated to adjust for room acoustics in order to get the proper balance in tonal quality. These same amplifiers and compensator are used when composite prints are projected for screening. The projector for the re-recording room is a standard Brenkert projector equipped with a preview attachment and a multiduty motor, and it does double duty for interlock shows to clients and as a fourth film phonograph when needed—using the sound head without having to thread through the picture aperture. This sound head is the only one not equipped for push-pull operation, the other three having that feature.

The output of the low-pass filter is bridged and the signal divided to permit recording on other systems simultaneously or for re-recording monitoring. The magnetic recorder is normalled from this bridging coil to receive whatever signal the optical recorder receives. Some engineers may feel that the magnetic should be fed by a signal free from high- and low-pass filters, to take advantage of its inherently fine frequency response, but here it is felt that

[*Continued on page 64*]

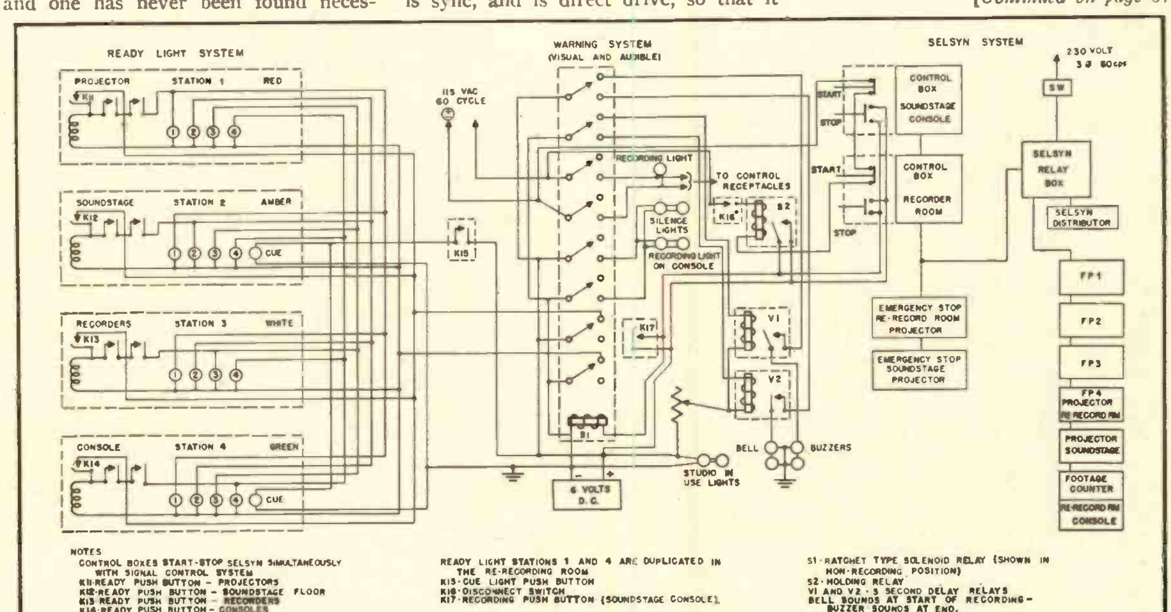

Fig. 2. Signal light and motor control system for small motion picture sound plant.

Handbook of Sound Reproduction

EDGAR M. VILLCHUR*

Chapter 8. Elements of Fidelity in Sound Reproduction

A discussion of the characteristics of a reproducing system which must be considered in evaluating the performance in terms of its fidelity.

THE CHARACTERISTICS of a (monaural) reproducing system that determine its ability to reproduce sound faithfully may be categorized as:

Frequency response
Transient response
Harmonic distortion
Intermodulation distortion
Power capability
Noise level
Dynamic range

Frequency Response

Frequency response is the most widely discussed factor in sound reproduction, and is often the first technical expression that the audio novitiate learns, although he is frequently led to accept a very incomplete meaning for the term. The frequency response of a system refers to the relative amplitude with which sounds or components of sound, varying only in that they are of different frequency, are reproduced. This response is typically such that a certain band of the frequency spectrum is reproduced with a given degree of uniformity, with output beyond the ends of the band dropping off. When the drop is sharp it is called *cut-off;* when it is gradual it is referred to as *roll-off.*

Thus there are two aspects of frequency response: the range of frequencies covered by the reproduced band with a specified minimum response, and the type and degree of variation which occurs within the band. Except where the range is very inadequate, the second of these aspects is usually the more important. So-called frequency response ratings which merely record the two frequency extremes of the reproduced spectrum may have little relationship to the quality of sound to be expected. Two loudspeakers with the same response at 40 and 15,000 cycles may produce entirely different tonal qualities due to the dips and peaks of acoustical output at less extreme frequencies.

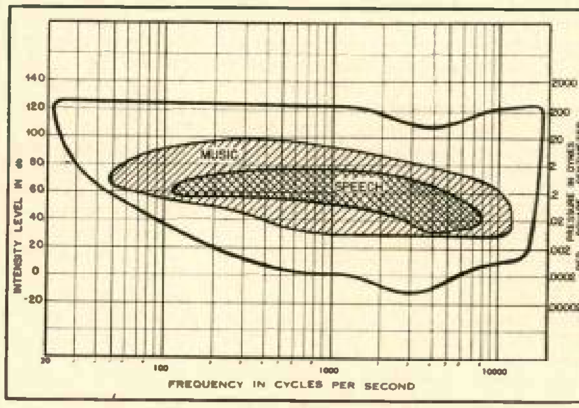

Fig. 8—1. Frequency and intensity ranges of speech and music. The solid line represents the limits of normal hearing. (Courtesy Bell Laboratories Record, June, 1934)

Uniform or "flat" response over the band of reproduced frequencies is desirable when two conditions are satisfied. These are: (1) that the reproduced sound is at the same intensity level (produces the same number of acoustical watts per square centimeter) as the original sound at a normal listener's position, and (2) that acoustical conditions in the room in which the sound is reproduced create more or less the same frequency discrimination as conditions in the original hall. When these two conditions do not exist, as is often the case, a rising bass characteristic to compensate for the Fletcher-Munson effect (see Chapter 6), and some tonal adjustment for discriminatory acoustical conditions should be provided by variable tone controls.[1]

We have seen that intensity differences on the order of a fraction of a db are just discernible at most frequencies and intensity levels, but this perceptive sensitivity is for single, pure tones heard under laboratory conditions. The degree of non-uniformity of frequency response that can be tolerated without effect on the quality of reproduction is undoubtedly somewhat greater, probably on the order of plus or minus one to two db. The indicated standard for maximum variation in frequency response is easily met in electronic circuits, more difficult to achieve in pickups, and, up to the present, an impossible ideal for loudspeaker systems, the very best of which reproduce their spectra with a variation of plus or minus five db. When sections of the frequency band are reproduced with ten db greater amplitude than others, tonal coloration must be expected.

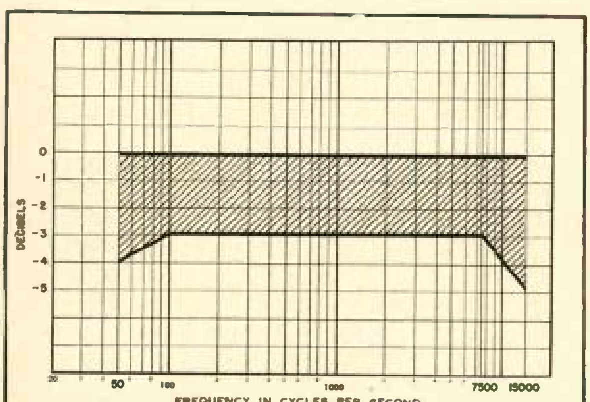

Fig. 8—2. FCC limits of frequency-response variation permitted FM broadcast stations (corrected for pre-emphasis). The range from 100 to 7,500 cps is restricted to a variation of ±1.5 db, while the standards beyond this band are not as strict. (The minimum frequency response standard for AM stations is 100-5,000 cps, ±2 db.)

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[1] The tone controls of an amplifier also provide compensation for associated equipment and program characteristics, but this discussion is limited to the desirable frequency response of the complete system.

The range of frequency response required to reproduce orchestral music without such coloration, and to include all significant overtones, transient effects, and noise, is not a matter for theorizing but for experimental verification. We are not concerned with reproducing all of the acoustical energy of the orchestra for its own sake, but only that part of it which has significance in the perception of quality.

Figure 8-1 is a chart reproduced from the *Bell Laboratories Record*, which includes information on the frequency ranges required for the reproduction of music and speech. This chart agrees closely with the data appearing in Snow's graph of audible frequency ranges for speech and music (see Chapter 5). We may consider the frequency range for perfect apparent fidelity of speech reproduction to be from 100 to 8,000 cps (Snow uses a slightly higher upper limit for female speech), and the range for perfect apparent reproduction of orchestral music as 40 to 15,000 cps.[2] A study in detectable band-width differences[3] has indicated that this last upper limit must be reduced to about 11,000 cps before the change becomes noticeable. It may be inferred that a corresponding liminal (just noticeable) change in the bass limit would probably be to 60 cps or higher. These latter figures may therefore be taken as reasonable standards for a high-fidelity reproducing assembly one step down from perfection as far as frequency range is concerned.

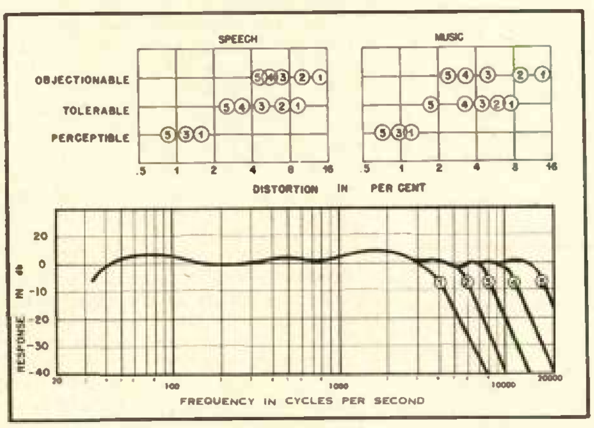

Fig. 8—4. Objectionable, tolerable (for low-fidelity units), and perceptible total rms harmonic distortion in music (produced by triode), for various frequency cut-offs. From Olson, Elements of Acoustical Engineering; courtesy D. Van Nostrand Company, Inc.

The only standard orchestral instruments whose fundamental frequency ranges go below 40 cps are the piano, the harp, the pipe organ, and the contra bassoon. It has been experimentally established that the fundamental energy in the very low notes of the piano is so small as to have no effect on perceived quality, and it is logical to assume that the same is true of the harp. The extreme low-frequency contributions of the two other instruments will rarely be a factor in the tonal value of reproduced music. The fact that there are a few organs in the world which produce subsonic 8-cps fundamentals should not unduly influence the design of a reproducing assembly made for home entertainment.

Sometimes too great a concern with extending the frequency limits of reproduction to extreme values has led to rationalizations of the need for such extension. It has been stated, for example, that sound reproducing equipment must be capable of transmitting supersonic and subsonic sums and differences between the frequency components of music, or that a band-pass extending into the subsonic region is necessary to reproduce properly a sound whose intensity is changing at a subsonic rate. We have seen that sum and difference frequencies of an intermodulatory nature can only exist when the signal is passed through a non-linear device such as the ear; whatever such intermodulation products will be formed in the ear of the concert listener will also be formed in the ear of the listener at home, without direct transmission of these products. As for variations of sound intensity at subsonic frequencies produced by vibrato, beats, or other causes, such effects are not tones in themselves but amplitude modulations of the wave envelope. Just as the coupling circuits of broadcast AM receivers transmit the wave envelope of the r.f. carrier without having any direct response to the audio frequencies involved, the audio reproducing system does not require subsonic response in order to transmit and reproduce amplitude variations of subsonic frequency. A mathematical analysis of the varying signal finds the modulating frequency conspicuously absent.

The frequency-response range of stages of the reproducing system within a feedback loop must be extended far beyond the audible limits in order to prevent phase shift and regeneration. Phase shift in the overall system, however, has little effect on perceived quality (see Chapter 6), and is, in any case, normally introduced by the reactive elements of tone controls or equalization networks.

Figure 8-2 shows the minimum frequency-response range and maximum

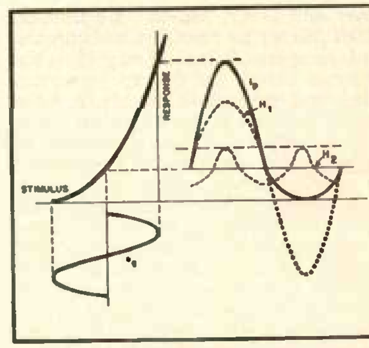

Fig. 8—3. Generation of second harmonic distortion by a non-linear transfer characteristic. The actual output, i_p, may be analyzed into fundamental and second harmonic components, shown as H_1 and H_2. e_g is the input signal. (After Reich, Theory and Applications of Electron Tubes; courtesy McGraw Hill Book Co., Inc.)

variation allowed FM broadcast stations by the FCC.

Balance

Tonal balance refers to the symmetry of frequency response deficiencies relative to the mid-point of the audio-frequency spectrum. Since there are an equal number of useful octaves on each side of the geometric mean, which is about 800 cps. this frequency constitutes the perception mid-point of the spectrum.

It is generally considered desirable to balance losses in the treble region with more or less corresponding losses in the bass, and vice versa, even though some sacrifice from absolute fidelity is involved. This view may be explained by an analogy in the field of color reproduction of pictorial material. Inadequate reproduction of the cold blues and greens, giving a picture a warm reddish cast, could be offset by a corresponding reduction in the intensity of the warm oranges and reds, a reduction which, while it involved a loss of absolute fidelity, might serve to give the picture a less artificial character.

Aural balance is often discussed in terms of the relationship between the

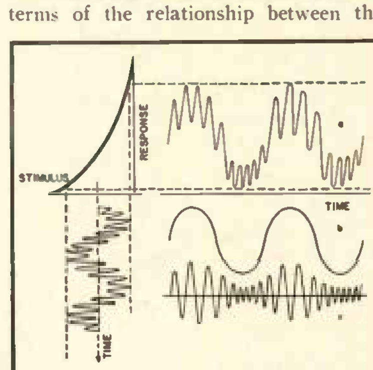

Fig. 8—5. Intermodulation produced by non-linear transfer characteristic and attendant distortion. (a) is the output signal, (b) and (c) are the distorted low-frequency component and the amplitude modulated high-frequency component, respectively. From Reich, Theory and Applications of Electron Tubes; courtesy McGraw Hill Book Co., Inc.

[2] These are the figures used in *Musical Engineering*, by H. F. Olson.

[3] D. K. Gannett and I. Kerney, "The discernibility of changes in program band width," *Bell System Technical Journal*, Jan. 1944.

upper and lower "cut-off" frequencies, which this writer considers unfortunate. Such an approach is valid only when the response curves are flat up to certain points and then cut off sharply. A more typical condition involves gradual losses over a fairly large band at the end of the spectrum, which would be balanced by similar losses symmetrical (on a logarithmic scale) to the mid-point. An additional difficulty in achieving balance by selection of cut-off frequency is that a change of frequency range involving only the bass extreme of the spectrum will leave the general character of the sound unchanged during the major part of most music, when there is no significant energy present in this frequency region. One conclusion about which there seems to be agreement, however, is that there is a large amount of latitude permissible before the response becomes audibly unbalanced. Most tone control circuits are capable of making the necessary corrections.

All of the above discussion of frequency response refers to the effect on a listener at one particular position. It is important, of course, that the response be as uniform as possible over as wide an angle as is required to cover all listening positions.

Transient Response

Proper reproduction of the instantaneous wave forms associated with the starting and stopping of sound vibrations requires a much more extended high-frequency response than is called for by the steady-state tone. The initial percussive or plucked impulse contains many high-frequency partials which are not part of this steady tone and which disappear after a short time.

Inadequate high-frequency response deadens the quality created by transient components. But transients suffer from accentuated response as well. The existence of a resonant response peak increases the tendency of the system, when stimulated by an impulse (especially at or near the frequency of the peak), to continue to oscillate after the stimulating impulse has stopped. This tendency is directly proportional to the height of the peak and to the extent to which it is confined to a narrow band of frequencies, that is, to the "Q", and is inversely proportional to the degree of damping. The effect is called *hangover* in the bass; in the treble ranges it is often referred to as ***singing***. We thus see that transient response is directly related to both the range and the uniformity of frequency response.

Distortion

The output of a device is said to be distorted when the instantaneous response is not directly proportional to the instantaneous stimulus at all times. This distortion has several synonymous titles: it is called ***non-linear, amplitude***, or *harmonic*. *Figure* 8-3 illustrates how the wave form distortion produced by a non-linear transfer characteristic[4] may be analyzed into partials consisting of the fundamental and second harmonic.

The spurious harmonics that are created in this way do not have as great a direct irritation value as might be supposed. They are, after all, harmonically related (in a musical sense) to the fundamental, and they may serve to intensify or cancel natural harmonics which already exist. Yet harmonic distortion is very unpleasant, and we are sensitive to very small amounts of it, as indicated in *Fig*. 8-4.

The explanation lies largely in an *effect* of harmonic distortion called intermodulation, which was touched upon in the discussion of subjective tones. When a complex tone consisting of both low- and high-frequency components is passed through a device with the transfer characteristic of *Fig*. 8-3, the high-frequency signal will be reproduced with increased amplitude during most of the first half of the cycle, and with reduced amplitude during most of the second half of the cycle. This is because during these periods the device has, in turn, exaggerated and diminished response to all stimuli. The amplitude modulation created by this non-linear transfer characteristic is illustrated in *Fig*. 8-5.

Certain wave forms, such as those produced by combinations of fundamentals and harmonics, are easily derived graphically and understood intuitively. Others, more complex, do not have components that appear quite as obvious. For example, the wave form that results from combining, in a linear system, signals of 940 cps, 1000 cps, and 1060 cps is a 1000-cps wave, amplitude modulated at 60 cps. It is a little difficult to see 940 cps and 1060 cps in this last wave form, but an application of the principles of beat phenomena will indicate that every one-sixtieth of a second the phase of the 940-cps signal will have advanced, and the phase of the 1060-cps signal will have retreated, by 360 deg. relative to the 1000-cps signal. The three signals thus return to the same phase relationship every sixtieth of a second, and a 1000-cps wave pulsating in amplitude at the rate of 60 cps implies the existence of the two sum and difference frequencies, which are called ***sidebands*** in r.f. parlance. (It must be emphasized that this modulated wave form does ***not*** result from mixing 60 and 1000 cps in a linear system.[5] The reference to beat phenomena is used here to indicate how an amplitude modulated wave implies the existence of sideband components, and not to explain intermodulation, which is ***not*** a beat phenomenon.)

When the amplitude modulated wave form of *Fig*. 8-5 is created through non-linear distortion, therefore, two new sideband frequencies are also created. If the input signals of *Fig*. 8-4 are 60 and 1000 cps,[6] the output will contain, in addition to these two frequencies, the intermodulation products 940 and 1060 cps. Both are discordant to 1000 cps.

When the amplitude distortion is such that the positive and negative halves of the cycle are symmetrical (odd orders of harmonics), the complete

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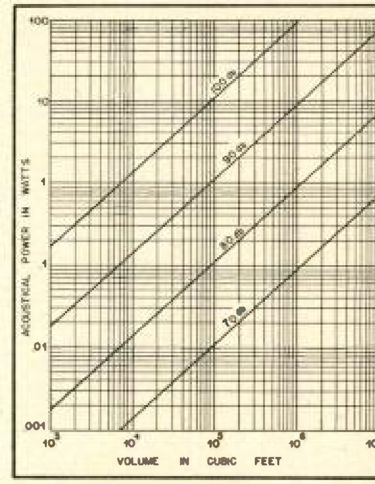

Fig. 8—7. Acoustical power required to create saundand intensity levels of 70 to 100 db. To convent to electrical power, divide by efficency of the loudspeaker.

AM STATIONS

MAXIMUM HARMONIC DISTORTION AT LESS THAN 84% MODULATION	MAXIMUM HARMONIC DISTORTION 85% TO 95% MODULATION	MAXIMUM HUM AND NOISE, 150-5000 cps	MAXIMUM HUM AND NOISE, OTHER FREQS.
5%	7.5%	50 db BELOW MAX. AUDIO OUTPUT	40 db BELOW MAX. AUDIO OUTPUT

FM STATIONS

MAXIMUM HARMONIC DISTORTION 50 - 100 cps	MAXIMUM HARMONIC DISTORTION 100-7500 cps	MAXIMUM HARMONIC DISTORTION 7500-15000 cps	MAXIMUM HUM AND NOISE, 50-15000 cps
3.5%	2.5%	3%	60 db BELOW MAX. AUDIO OUT.

Fig. 8—6. Maximum harmonic distortion and noise permitted broadcast stations by the FCC.

[4] A transfer characteristic is a graph which plots instantaneous response against instantaneous stimulus. For example, the transfer characteristic of a vacuum tube is normally plotted as output plate current *vs*. stimulating grid voltage; of a pickup, output voltage *vs*. stimulating velocity, etc.

[5] It is this fact which makes it necessary to operate the heterodyning "mixer" stage of a super-het receiver in Class C.

[6] These two frequencies are harmonically related; 1000 cps is the tenth harmonic of 100 cps, which is separated from 60 cps by a musical interval of a major sixth.

The Great Loudspeaker Mystery

H. A. HARTLEY*

An exercise in Metaphysical Philosophy

A YEAR OR TWO AGO there appeared in a certain British technical journal an article by a loudspeaker manufacturer concerning the design of loudspeaker diaphragms. In it, there appeared the old platitude that the perfect diaphragm was a rigid piston of negligible mass. From this premise it was argued that since the perfect diaphragm could not exist, attempts should be made in practical work to approach this perfect concept as closely as possible. Therefore . . . and so on.

Now it so happens that I have never been particularly attracted by this notion of the perfect diaphragm, so I wrote a letter for the correspondence columns of the journal in question, the burthen of which was that the idea that the perfect diaphragm should be an infinitely rigid piston was nonsense, because, I went on to say, if the diaphragm were infinitely rigid it could produce only one frequency at a time—in other words it could reproduce only a sine wave. The editor's technical adviser wrote a private letter to me to ask if this letter of mine was a leg-pull, for if they were to take me at my word, the publication of my letter would make me the laughing stock of the audio industry of Britain. I said that it wasn't, and that I was quite ready to stand by the consequences of my letter being published, and would they please publish it? They did not, and I have never been able to determine whether it was done out of kindness to me or out of consideration for themselves. During a recent visit to New York I mentioned this incident to *your* editor, and he looked at me in a quizzical sort of way, not quite sure whether I was pulling his leg, or whether I had got something. I threatened him with a full thesis on the matter, and here it is. You have my assurance that it is a serious contribution to audio thinking, and I am prepared to abide by the consequences of putting my name to a major heresy.

Let us get the problem clearly stated. It is the conventional idea that the perfect speaker diaphragm should be an infinitely rigid disc, preferably of no mass. It is my idea that the perfect speaker diaphragm should be a flexible entity (shape not specified) which in its various parts vibrates in various ways so that the over-all effect is to reproduce all frequencies with constant output. The problem is therefore: Which is correct? I shall try to help you solve the mystery.

Logic vs. Mathematics

There has been some feeling that the articles in Æ should butter the popcorn. If this can be done without spoiling their authenticity I am all for it. Mathematical treatment of loudspeakers is not easy and can be so difficult as to be impossible in certain directions. I believe that my thesis can be comprehensive without being mathematical. Those readers who would like to have an extremely able exposition of the mathematical analysis of loudspeaker behavior are recommended to study the book "Loud Speakers" by N. W. McLachlan. I warn them that it calls for mathematical knowledge of a high order, and that is why I have considered my treatment should be rather one of non-mathematical logical thought. However, the technical statements I make are provable by rigorous mathematical processes.

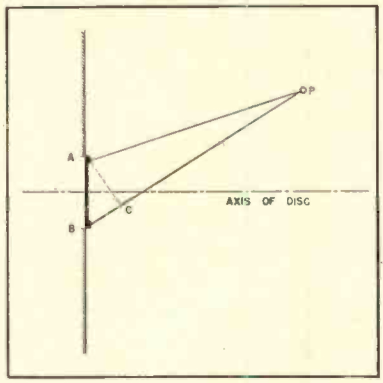

Fig. 1. Representation of a circular disc, AB, in a rigid baffle to show the cancellation due to phase distortion when the listener, P, is not on the axis of the disc.

My argument comes in the domain of metaphysics, and I hasten to add that metaphysics has nothing to do with mysticism. It is characteristically anti-metaphysical to suppose that this science of thought processes has anything to do with psychology, theology, or demonology. It is an exact science, concerned with the examination of thought, argument, and knowledge.

It is an axiom of metaphysical philosophy that argument based on opposing absolute presuppositions can never be resolved. If you and I start arguing in a general sort of way as to what is the most nearly perfect diaphragm for a speaker, and you say "The cone must be made as stiff as possible" and I retort that that is all wrong, we can argue about this until all is blue without getting any further because we hold conflicting presuppositions. Every statement made by a man is, consciously or otherwise, an answer to a question, and the question has arisen out of some earlier statement which in turn derived from another question. The metaphysician traces these statements and questions back to their source until he arrives at something which is not susceptible to logical treatment. That "bit of knowledge" when examined is found to be a matter of belief, not knowledge, and he calls it an "absolute presupposition" because it is the absolute source of your subsequent argument. He does not attempt to criticise the validity of your absolute presupposition, but he will unerringly point out any flaw in the logical development of your argument based on it.

On this basis let us consider the problem I have put to you. The statement that the perfect speaker diaphragm should be an infinitely rigid disc of no weight is an absolute presupposition, because it is an article of faith. How can it be otherwise? It is clearly impossible for any person to produce an infinitely rigid disc of no weight, so why call for an abstraction as the goal, except as an act of faith? But if we seek to find a *reason* for the common acceptance of a hypothetical device as the aim of all speaker designers, then the search is a short one, for the mathematicians will tell you that whereas analysis of the behavior of a disc *in vacuo* is comparatively easy, it has not yet been found possible to analyze the behavior of a cone. If, therefore, mathematical treatment is not available for this case it passes out of the realm of exact science, and there must be substituted something which can be analyzed mathematically, and that is the disc.

Now it can also be proved mathematically that under the impulse of a voice coil actuated by alternating currents the disc will distort in a way which can be exactly foretold without experiment. If, therefore, we are conscious that the disc in practice does distort, we easily pass on, or back, to the supposition that the disc, to be perfect, should not distort, and to achieve that it must be rigid. By the same process, but by different mathematical treatment, it can be proved that transient reproduction is differentially distorted the greater the mass of the disc, and the perfect diaphragm should also have no weight. And so you are back to your absolute presupposition, which, as I have pointed out, is not a matter of scientific knowledge but of faith. And I am entitled to disagree with your faith, even if I accept your facts.

But, I can imagine you saying, what am I getting at? On the one hand I admit that mathematics proves that a practical disc distorts, but a theoretical rigid disc would not distort, and mathematics also proves that a practical disc having mass also distorts, and by inference a theoretical massless disc would not distort. On the other hand I am obviously not satisfied with the idea that the perfect disc, infinitely rigid and infinitely light, which can be mathematically proved not to distort, is the ideal to which, in an imperfect world, we can only approximate to. Am I just being difficult? The statement made above would probably be acceptable without careful thought, but it contains two serious flaws in logical argument.

First is the assumption that if it can be mathematically proved that a non-rigid disc must distort, it follows that a rigid disc will not distort; and that a disc with mass hav-

* 152, Hammersmith Road, London W. 6, England.

ing been proved to distort transients, it follows that a massless disc will not. The second is the assumption that there is no other form of distortion accompanying the use of a rigid massless disc. Both these assumptions are wrong. as can be proved mathematically. Let us take only two examples of the latter's wrongness.

In *Fig.* 1, AB represents a rigid circular disc mounted in an infinite rigid baffle. P is a point off the axis of the disc, a position such as might be taken by any listener to a speaker. Now P is nearer to A than to B and the radiation from B will be out of phase with that from A by an amount $2\pi (BC)/\lambda$, where BC is the difference in the two distances and λ is the wavelength of the radiation. At low frequencies BC is small compared with λ, and the phase difference is negligible; but at high frequencies λ may be larger than BC with consequent considerable phase distortion. Specifically, if $BC = \frac{1}{2}\lambda$ the radiation from A and B will be in opposite phase, and cancellation of the radiation will occur.

This can be translated into the form of a polar radiation curve. If the phase difference effect is not noticeable at low frequencies it can be assumed that the polar curve for a low frequency would be semicircular, and that the radiation would be hemispherical; mathematically treated it can be shown that at, say, a frequency of 4000 cps the polar curve will be an ellipse and the radiation ellipsoidal. Worse, the larger the disc the narrower the ellipse; and a little reflection will show that this must be the case, since the difference between PA and PB will be greater, the larger the disc. Since it is the purpose of a loudspeaker (and its diaphragm) to radiate sound as well as possible, one can naturally suppose that the radiation should be hemispherical at all frequencies, but we have just seen that the rigid disc is inherently incapable of achieving this.

The Flexible Disc

If the mathematical analysis be carried some stages further to examine the behavior of a flexible disc, one which develops nodes under the impulse of an electric current applied to the voice coil, we find that the polar curves at high frequencies are much better than those of a rigid disc, and again some reflection will indicate why this should be so, for in my figure B may be on a node and A at an antinode, thus reducing the difference between PA and PB.

Our metaphysician may now be allowed to say a few more words. He did not quarrel with your absolute presupposition that a rigid massless disc was the theoretically perfect diaphragm, because he showed that that was a matter of belief, and you are entitled to believe what you like; but you are now under an obligation, if you insist on adopting the scientific way of thinking, to find out if there are any grounds for your belief, and I have just shown that there are not. I am not prepared to accept the excuse that you *believe* your perfect disc is the best *compromise,* for science has no room for compromises—a thing or a thought either is or it isn't. But it has been truly pointed out by no less a savant than the late Professor R. G. Collingwood that people are very touchy about their absolute presuppositions, so I shall not labor the point. I shall content myself by observing that the belief that a massless disc is the perfect diaphragm has no justification either in theory or practice.

And now, if I may, I should like to put my second cat among the pigeons—my argument that if the perfect diaphragm, the infinitely rigid disc of no weight, could be made, it could only reproduce a sine wave. I am afraid that when we enter this field of speculation we cannot have confirmation from mathematics. for the problems are so complex as to defy analysis. Our speculation can only be philosophical, and we can only accept what appears to be the more reasonable theory. Let us examine this notion of mine.

A rigid disc is inflexible. If vibrations can be transmitted through it then it is flexible. Since it is rigid it can only act as a pure piston incapable of being deformed. Apply a sine-wave to it through the medium of the voice-coil and former of any frequency you like, say 100 cps. The disc, being a suspended diaphragm, vibrates to-and-fro at that frequency. Now, while continuing to cause it to vibrate at 100 cps, you also apply another frequency of, say, 200 cps. What will happen? Since it is your supposed perfect diaphragm it will reproduce both frequencies at amplitudes strictly proportional to the amplitudes of the applied currents. but how can it do this? Since it is infinitely rigid it cannot vibrate in one part at 100 cps and in another part at 200 cps so the only possible theory is that it moves forwards and backwards non-sinusoidally, the form of its movement being the counterpart of the harmonic sum of the two separate frequencies. This also involves the acceptance of the consequence that the movement of the diaphragm includes accelerations and decelerations with each half-cycle. I agree that if the diaphragm has no mass it could perform in this way, but only *in vacuo,* for in air the air friction would be enough to cause distortion of the necessary accelerations and decelerations. This implies that the definition of the perfect diaphragm being an infinitely rigid disc of no mass must also be restricted to the condition *in vacuo,* but this makes nonsense of the whole idea, for without air we cannot hear sound. and without sound output the perfect diaphragm is useless.

But the perfect piston protagonist has another headache coming. We have seen that his belief necessitates the diaphragm moving forwards and backwards in a non-sinusoidal manner exactly reproducing the non-sinusoidal current applied to the voice-coil. Forgetting the *in vacuo* restriction, a fatal snag, we have to envisage the piston moving forwards and backwards reproducing the fantastically complex waveform of a full orchestra by vibrating to-and-fro with hundreds, possibly thousands, of accelerations and decelerations in a tiny fraction of a split second. Is this conceivably possible? And if it is, how can it, having escaped the air friction, transmit this extraordinarily complicated waveform to an elastic medium like air, which will initially absorb all the fine detail long before it has a chance of being propagated? As we must expect the perfect diaphragm to reproduce all the frequencies applied to it, it is clearly impossible for it to do so by complex oscillation to-and-fro, so whether my statement that the infinitely rigid disc could only produce a sine-wave is true or not, the chances are overwhelmingly in its favor that it is true and it certainly cannot be proved wrong. We have also seen that the rigid disc cannot be perfect by its very nature, so it seems time that the whole of this absolute presupposition can be jettisoned. It follows that all the theoretical ideas based on this presupposition must also be thrown out. In short I am constrained to follow the late Henry Ford's example when he spoke of history and say that all conventional loudspeaker design of a theoretical nature is bunk.

"Design" of Speakers

How. then, you may well ask, how *does* one design loud speakers?

I invite you now to consider *my* absolute presupposition, which is that the perfect diaphragm is a flexible entity which vibrates in various sorts of ways so that the sum total of its performance is to reproduce all the frequencies imparted to it by the current applied to the voice-coil by imparting movement to various zones of air so that the ultimate sound-waves impressed on the listener's ears create a sensation exactly similar to that created by listening to the original performance. This rather complicated sentence carries within it an inferential absolute presupposition—that you cannot design speakers by any mathematical process, since analysis of the process is beyond human knowledge. I have already explained how even high-grade mathematics can only undertake what are really the simplest phenomena, and then only suppositional cases, for, in general, the formulas developed presuppose conditions which are not met with in actual practice.

There is a good deal of justification for accepting my absolute presupposition as an article of faith (for it is nothing more) and using it as a basis for a logical development of argument. Consider, for example, a musical instrument such as a violin.

It is an ordinary commonplace scientific fact that the fundamental frequency of the note produced by drawing the bow across the strong is a function of the length of the string between the bridge and the performer's applied finger. Whether the performer is a virtuoso or a tyro, provided the distance is *x* inches the frequency of the note will be *y* cps. But in practice the *quality* of the note (of given frequency) depends on at least two things, one the way in which the bow is drawn across the string, the other the nature of the body or belly of the instrument. It requires no great musical knowledge to be able to hear the difference between the sawing of the beginner and the polished perfection of the great violinist. It is not just a matter of producing the right notes at the right moment—sheer practice in manual dexterity will in time produce this result—but more important is the nature of the sound produced, and that depends on bowing. But the expert is not content with this; he will try to obtain a specimen of the work of an Amati or a Stradivari so that his fingering and his bowing will not be spoiled by poor instrumental tone. When all this has been

[*Continued on page 53*]

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The Protection of Ideas

ALBERT WOODRUFF GRAY*

Legal aspects of creative thought are also important to engineers.

THE SUGGESTION for an improvement in radio cabinets, accompanied by sketches of the proposed innovation was submitted to a radio manufacturer by a firm of cabinet designers. Later the sketches were returned by the manufacturer with a letter that it was not interested but should any of the designs be used, payment would be made on a royalty basis of 1 per cent of the cost of the cabinet, to be paid by the cabinet manufacturer.

The sketches submitted by these cabinet designers to the radio manufacturer had no earmarks by which the designers might identify the plans as their own. When later these same features appeared in a cabinet produced by this manufacturer, suit was brought by the designers for the value of the plans they had submitted.

The defense of the company, which the court sustained, was that the design incorporated in these cabinets was conceived by the president of the company from the suggestion of an automobile hood and that at no time had he ever had any knowledge of the ideas or suggestions submitted by these designers.

Nearly two hundred years before this incident an English judge said in the decision of a case involving the exploitation of ideas,

"Nothing can be the object of property that is not capable of distinguishable proprietary marks. The principle of this first institution of property was established to preserve the peace of mankind which could not exist in a promiscuous scramble. Therefore a moral obligation rests on all that none should intrude upon the possessions of another.

"But this obligation could only take place where the property was distinguishable and everybody knew that it was not open to another. The breach of a duty must be willful to make it criminal. It was necessary therefore that every person should have some indicia, some distinguishing marks upon his property to denote his being the proprietary, for hard would be the law that should judge a man guilty of a crime when he had no possibility of knowing that he was doing the least wrong to any individual.

"Now where are the indicia or distinguishing marks of ideas," asked that eighteenth century English judge. "What distinguishing marks can a man fix upon a set of intellectual ideas so as to call himself the proprietor of them? They have no earmarks upon them, no tokens of a particular proprietor."

There was lacking in these radio cabinet designs the earmarks or indicia, mentioned in that English decision, that would have aided them in the support of their claim for compensation, had the ideas for this innovation been submitted under the provisions of an agreement or contract between themselves and the radio manufacturer.

This lack of means of showing the identity, the absence of the earmarks, in circumstances of this character, was the subject of a comment by the Court of Appeals in New York state, in denying recovery to the proprietor of another idea claimed by him to have been wrongfully appropriated by another.

"Without denying that there may be property in an idea or trade secret or system," said that court, "it is obvious that its originator or proprietor must himself protect it from escape or discovery. If it cannot be sold or negotiated or used without disclosure it would seem proper that some contract should guard or regulate the disclosure, otherwise it must follow the law of ideas and become the acquisition of whoever receives it."

Idea Outlined in MS.

An action before a New Jersey court involved a manuscript in which had been outlined a valuable idea but with no provision for compensation to the originator. The plan was adopted. Later when an action was brought to recover the value of this plan the court said of the recovery claimed by the author,

"He has undoubtedly the right to claim protection for his manuscript. The combination of words of which it is composed is also protected. The law has never attempted to go beyond this and to enjoin for the benefit of the author, after publication, the use of ideas contained in his work. In the case of secret processes of manufacturing the law does to a certain extent enjoin the use of ideas. In enjoining the use of formula it restrains the wrongdoer from putting the ideas formulated to practical account. The protection ends when the secret becomes known.

"If the idea contained in a patented device of "A" suggests to the mind of "B" another idea that would not have arisen in the mind of "B" but for the stimulus of the prior idea, "A" can claim no property in that, and yet "B" has mentally appropriated "A's" idea and made it the basis of his own; and I do not suppose it has ever been contended that the entire public are not at liberty to subject "A's" idea to such investigation and discussion as it may desire.

"It was, as far as its originator was concerned, an idea, pure and simple. Now it has never in the absence of contract or statute, been held, as far as I am aware, that mere ideas are capable of legal ownership and protection."

In a later opinion rendered in this same litigation the New Jersey court made this supplementary comment, "Undoubtedly ideas, if valuable or even thought to be valuable, may be the subject of bargain and sale. They may be the subject of contract but they must be protected by contract."

An idea once released by publication freed for the use of whomsoever may so wish, like a wild animal freed from its cage, can be the property of no one. Substitute for a cage a contract and for the bird or beast of the outdoors, an idea, and the analogy is complete. In every instance there must be a contract, either express or implied. Lacking that the idea is irrecoverable when once it has gone from its owner.

An idea was submitted to a manufacturer who had made no agreement for payment and the outline of this idea was unaccompanied either by drawings or the findings of experimental work.

If the manufacturer derived benefit from the idea, asserted the federal court denying to the originator of this idea a recovery, it is nevertheless not indebted to this originator because it did not offer to or make any agreement to pay for such mere suggestions as the possessor of this idea made. When he voluntarily divulged his mere idea and suggestion, whatever interest he had in it became common property and was available to the manufacturer.

A contract protecting an idea against exploitation and loss must obviously precede the disclosure of the idea itself. The idea, however, must be thus protected in its disclosure but it must be novel to the recipient and of a substantial value to him.

"When information is proffered as consideration for a contract it is necessarily implied—it is indeed the essence of the proffer that the information shall be new to the one to whom it is proffered," said the Connecticut Supreme Court in its decision of a case of this character.

"A statement to one of what he already knows is not, as to him, information but merely a statement of fact already known. The imparting of information in a situation like this must involve an active process, resulting in arousing or suggesting ideas or notions not before existent in the minds of the recipient, otherwise it is not information in the true sense of the term, although it may be a statement of fact."

In Indiana some years ago an idea or scheme was submitted with a letter that said, in substance, that the plan was submitted for approval. This was followed by an outline of the scheme and the letter concluded with, "I trust that this idea

[Continued on page 51]

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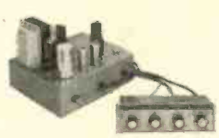

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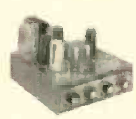

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The Audio Fair—Los Angeles

Well on its way toward topping the New York Audio Fairs in attendance and number of exhibitors, this newest of showings will play host to many who have been unable to attend the Eastern exhibits.

IT IS SAID about one state in the Union that whatever anyone else does, they do it better. And it is beginning to appear that this same statement might also be applied to the audio industry in Southern California, for the first Audio Fair—Los Angeles seems slated to come a very close second to the 1952 Audio Fair in New York —if it doesn't top the Eastern event, which it could well do.

February 5, 6, and 7 are the dates, and the Alexandria Hotel in Los Angeles is the place. The entire audio industry will supply the "who" unless the attendance should be entitled to that distinction. And those who have ever attended an Audio Fair know what is in store for the Alexandria for those three days—never a dull (or quiet) moment.

The Alexandria Hotel — one of Los Angeles' older and most aristocratic hotels of earlier days—is admirably suited for the Audio Fair. The rooms are large and the ceilings are high, which offers ideal listening conditions for audio equipment. The hotel is located in downtown Los Angeles, and is easily accessible from any part of the great metropolitan area.

The exhibits will feature custom high-fidelity television, home-music systems, and associated components, along with stereophonic and binaural demonstrations, and will be open to the public at no charge. According to Cap Kierulff, publicity director for the Audio Fair—Los Angeles, approximately 100 exhibitors will demonstrate their latest products which are now coming off the production lines to meet the increased consumer demand for higher quality radio-phonograph reproduction and for custom-built home entertainment centers. Exhibits will be open from 1:00 p.m. to 9:00 p.m. each of the three days, and will be located on the 6th and 7th floors of the hotel.

Hi-Fi Installations

Engineers, architects, contractors, and the home owner will see demonstrated the manner in which record changers and radio-phonographs, with their associated loudspeakers and cabinets, can be integrated into the design of walls, ceiling, and furniture; how all electronic equipment and components, including cable, can be installed to match the decor of any room, yet maintain an ease of accessibility for rapid servicing.

Educators and industry will see and hear the latest in public-address control facilities. Pre-recorded educational tapes, reproduced on the latest commercial and professional tape recorders, will be in actual operation, as will be new designs of record and transcription players, microphones, FM-AM-TV tuners, remote-controlled amplifiers, and corner loudspeaker enclosures.

Evidence of the trend toward high-fidelity reproduction of recorded and broadcast programs in the home, office, school, and club is reflected in the fact that the recent Audio Fair in New York drew over 13,000 spectator-listeners. Further evidence is in the fact that high-fidelity phonograph records have found a national market which is being well serviced by thousands of specialty dealers from coast to coast—catering to more and more consumers who desire near perfection in audio reproduction equipment, the only means by which they can achieve true listening pleasure.

Some of the members of the Industry Advisory Committee list the qualifications of Gina Lund, candidate No. 1 for the title of Miss Audio Fair—Los Angeles. Left to right—Robert Newcomb, James Pelham, Miss Lund, Bill Thomas, Boyd McKnight, and Fair Manager Bill Cara.

Industry Advisory Committee

Recognizing the potential growth of High Fidelity Audio for home sound reproduction, youthful William L. Cara organized an industry Advisory Committee of some 14 members, and has developed the Audio Fair—Los Angeles into an attraction which will satisfy the growing need for the consumer and dealer to investigate all available top-quality audio equipment under one roof —with the added advantage of being able to *hear* as well as see the equipment in operation.

The Advisory Committee is composed of Boyd McKnight, chairman of the Los Angeles Section of the AES; manufacturers Robert Newcomb of Newcomb Audio Products Co. and William Thomas of James B. Lansing Sound, Inc.; Representatives George Davis, Richard Hastings, Frank Koessler (Neely Enterprises), Lee Owens, and George Tivy; distributors Harry Braverman of Universal Radio Supply Co., James Pelham of Figart's Radio Supply Co., and Harry Shaffer of Hollywood Electronics; and publicity director Cap Kierulff, of Kierulff Sound Corp.

One feature of the show will be the selection of "Miss Audio Fair—Los Angeles." The accompanying photo shows candidate No. 1, Miss Gina Lund, with several members of the advisory committee.

Technical Sessions

The Los Angeles Section of the Audio Engineering Society has scheduled approximately 20 papers to be given during the three days of the Fair. Among them are the following, which are only representative of the subjects to be covered:

"FM Carrier Magnetic Sound System", by John T. Mullin, Bing Crosby Enterprises.

"Motor Drive Systems," by Olin L. Dupy, Metro-Goldwyn-Mayer Studios.

"Structure and Performance of Magnetic Transducer Heads," by Otto Kornei, Brush Development Co.

"Use of Electrical Delay Lines in Studies of Binaural Hearing," by R. S. Gales and R. G. Klumpp, U. S. Navy Electronics Laboratory.

"The Disc Recording Characteristic," by E. H. Uecke, Capitol Records, Inc.

"Aircraft Public Address System Operation Under High Level Ambient Noise Conditions," by J. C. Baker, Lockheed Aircraft Co.

"History and Development of Stereophonic Sound Recording," by R. H. Snyder, Ampex Electric Corp.

While the complete list of papers to be presented is not available at press time, Allan L. Wolff, chairman of the AES Papers Committee, West Coast, has indicated that there will be a full program of technical papers, and the Society plans to publish the entire text of all of the papers as soon as possible after the Fair.

The Los Angeles Section of the Society is sponsoring the show as part of its service to the industry.

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Making the Best of an Audio Transformer

N. H. CROWHURST*

A discussion of the effect of the various characteristics of a transformer on its performance. Curves are shown for low- and high-frequency performance of both input and interstage transformers.

The previous articles have clarified the properties of audio transformers bearing on its performance, and shown how the electrical properties of an individual specimen can be measured up. To make the best use of this information, charts have been devised to simplify prediction of performance, and adjustment of circuit values for optimum.

In any audio circuit, frequency response is usually considered first. Generally it is desirable that this should be as flat as possible for as wide a frequency range as possible, but sometimes deliberate narrowing of the band or correction for deficiencies elsewhere is required. The charts here given enable any of these requirements to be met with the minimum of effort.

Low-Frequency Response

Treatment of this end of the spectrum falls into two groups, as explained in the first article:[1] (1) direct-coupled transformers, either single ended or push-pull; and (2) parallel-fed transformers.

The treatment for direct-coupled transformers is simple, as previously explained. The response is generally of the form shown in *Fig*. 3. of the first article (reproduced here as *Fig*. A), the 3-db point on this curve being found by equating the reactance of the primary inductance to the total shunt resistive impedance, consisting of the input impedance or plate resistance of the tube. in parallel with the referred secondary load resistance and the resistive component of magnetizing current. To adjust this l.f. cut-off frequency, any of these impedances, or the primary inductance, may be altered to produce the desired result. Plate resistance can be modified by the use of feedback. The load resistance may be fixed, but in some applications, such as input and interstage transformers. resistance shunted across the secondary is not regarded as a load, and may be adjusted to suit response requirements. The primary inductance can be modified by adjusting the air gap, or by altering the d.c. polarizing through the primary, in the case of single-ended interstage transformers.

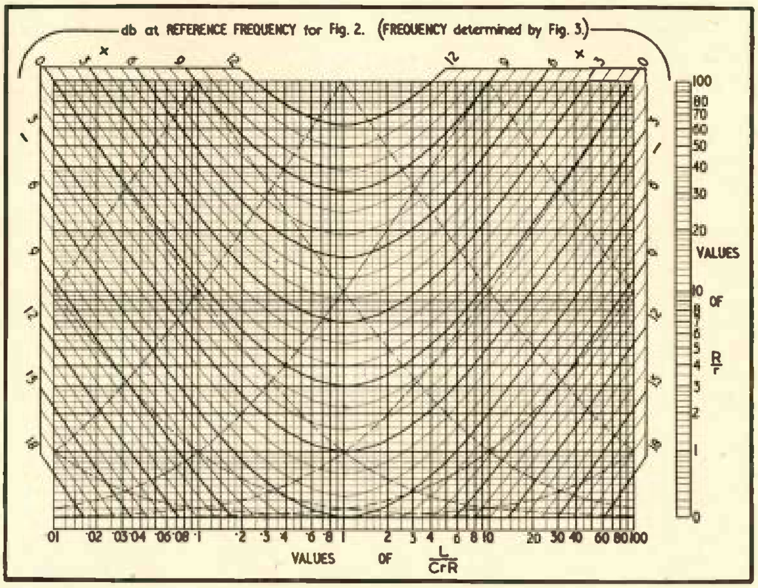

Fig. 1. Chart for determining response shaping from known values of L, C, r, and R (for significance of which see text); and for estimating effect of varying circuit values.

In direct-coupled transformers where d.c. polarizing is present, the inductance of the primary is reasonably constant, and the response curve follows the shape of *Fig*. A quite closely. But where there is no d.c. polarizing, and the transformer is cored up without an air gap, the primary inductance changes quite considerably with both amplitude and frequency, so the shape will not conform at all closely to this response. However, in these cases the l.f. response is usually made better than minimum requirements at all levels, so that its exact shape at any level is unimportant.

When using parallel-fed transformers the treatment is somewhat different. The l.f. cut-off network is analogous to a tuned circuit, and the shape of the response depends on the degree of damping present, series and shunt. Here again the primary inductance value is likely to vary with level and frequency, unless a component with a gapped core is used, so accurate prediction is rendered difficult. For this reason charts specifically for l.f. response prediction are not included, but those provided for h.f. response prediction are made adaptable for l.f. as well, where required. When using the charts for working out a suitable l.f. response, the frequency scale of *Fig*. 2 is reversed.

Although selection of conditions resulting in critical damping, or even a small degree of peaking, has been known, for the purpose of extending the l.f. range, it is not to be recommended for the reason that response will vary with signal level. On the other hand, when space or cost is restricted, it will be found that use of a coupling capacitor larger than an optimum value for critical performance, will deteriorate the l.f. response, rather than improve it.

High-Frequency Response

Variable inductances do not make the job so difficult at this end. Leakage inductance is constant, so quite accurate prediction is possible.

In *Figs*. 1 and 3, *r* always stands for the impedance connected to the low-impedance winding, and *R* for that connected to the high winding. So in step-up transformers, such as input or interstage, *r* is the input impedance or plate resistance in parallel with the coupling resistor, and *R* is the referred secondary shunt resistance.

* *82, Canterbury Grove, London, S. E. 27.*

[1] N. H. Crowhurst, "How good is an audio transformer?" Audio Engineering, March, 1952.

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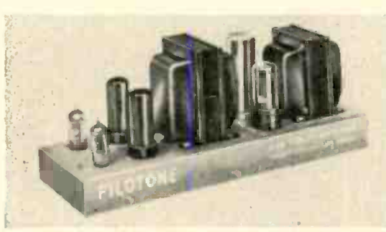

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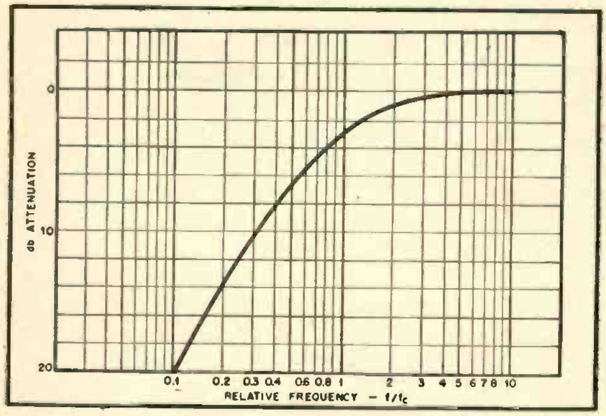

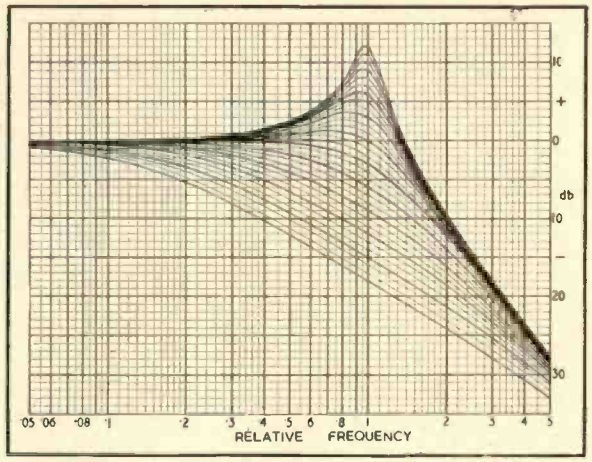

Fig. A. (above) Low frequency cut-off attenuation characteristic, 6 db per octave. Fig. 2. (right) Showing the possible variations of response shaping. The shape applicable to a particular case is determined by the Chart of Fig. 1.

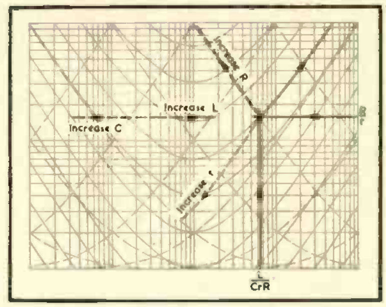

Fig. 4. Showing how the effect of changing circuit values is estimated by use of Fig. 1.

To apply the charts to l.f. responses, r is the parallel combination of the tube plate resistance and its coupling resistor, and R is the parallel combination of all shunt resistance components *after the coupling capacitor,* referred to the primary.

Output transformers employ a step-down ratio, so R becomes the source impedance of the plate circuit, and r is the load resistance referred to the primary.

For h.f. circuits, L is the leakage inductance referred to the appropriate winding, and C is the total shunt capacitance "seen" by the high winding, including its own self-capacitance, referred to the same winding as everything else.

For l.f. circuits, L is the primary inductance, and C the coupling capacitor value.

To use the charts, values of r and R are assumed, L will be fixed by the transformer itself, and C will have a minimum value fixed by the transformer and its associated circuit. From these values the quantities L/CrR, R/r, and LC are evaluated (L in henries, C in farads, r and R in ohms). The chart of *Fig.* 1 is then used to locate the shape of the response among those shown in *Fig.* 2, and the chart of *Fig.* 3 locates its position on the frequency scale. The right half of *Fig.* 3 applies to h.f. cut-offs and the left half to l.f. circuits. *Figure* 1 finds the db response at the frequency given by *Fig.* 3, which is marked 1 on the RELATIVE FREQUENCY scale of *Fig.* 2. This reference frequency is the point on the curve at which the slope is downwards at 6 db/octave.

Explanation of the reason for choosing this reference frequency will help to clarify the information conveyed by the charts. An alternative presentation[2] uses the peak frequency as reference, where peaking occurs, and an imaginary equivalent where there is no peaking, the exception being the boundary case between the two regions, which has to be treated separately. This presentation has the advantage of providing precision prediction of any individual response more readily, because a "universal" curve representing each region can be drawn, which is interpreted by its own db conversion chart to suit all values within the region. Its disadvantage for the present purpose is that the reference frequency slides off the opposite end of the scale as the critical boundary condition is approached, i.e. it falls to zero for h.f. cut-offs and rises to infinity for l.f. cut-offs; this makes presentation of varying response with different circuit values obscure, if this reference is used. The use of the 6 db/octave slope reference point enables continuous presentation through both regions.

The 6 db/octave slope point is also the frequency at which phase delay (or advance in l.f. cut-offs) is 90 deg. The phase characteristic is always symmetrical about this point, and the amplitude response may also be regarded as centered about this point, if it is referred to a 12 db/octave cut-off at this same frequency, instead of to zero level both sides of cut-off.

Notice that the curve in *Fig.* 2 with zero level at the reference frequency has a peak of about 1¼ db at a relative frequency of 0.7; the –3-db-at-reference-frequency curve is the critial or boundary case where peaking ceases; this is not quite the same as what is generally known as critical damping in connection with the introduction of transient distortion; critical damping, according to the accepted definition, is achieved by the –6-db-at-reference-frequency curve.

The scales used in *Figs.* 1 and 3 have been chosen to assist accuracy in drawing. The R/r scale on *Fig.* 1 is based on the law $\log(1+R/r)$, which allows a template to be used for the db rulings. By adjustment of the L/CrR scale, straight db rulings could have been used, but the conventional log scale makes reading of L/CrR values easier. In *Fig.* 3 the scale for R/r is based on log($1+$

[2] *Electronic Engineering* (England) Nov. and Dec., 1951, and Jan. and Feb., 1952.

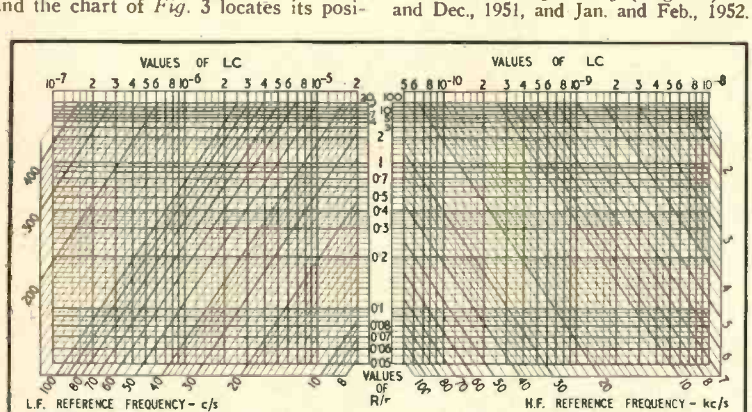

Fig. 3. Charts to determine the frequency location of the response predicted by Figs. 1 and 2. The left chart applies to l.f. cut-offs and the right chart to h.f. cut-offs.

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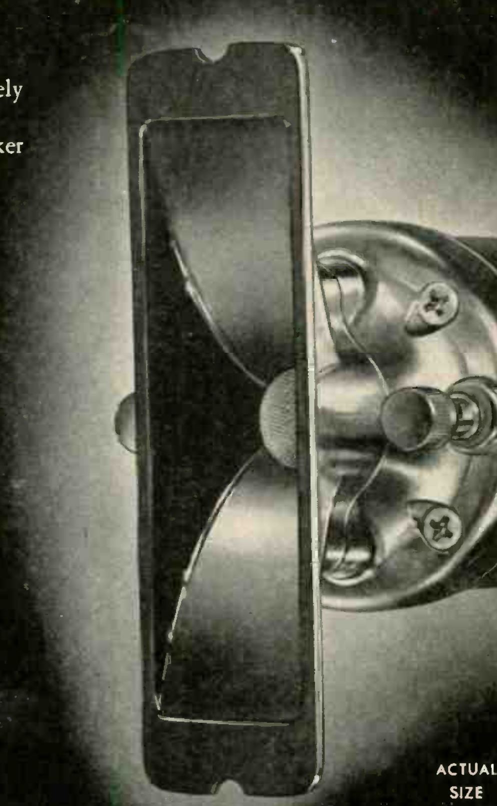

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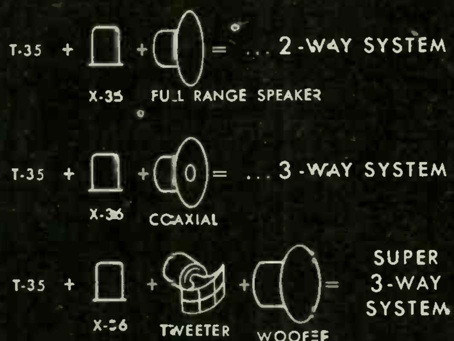

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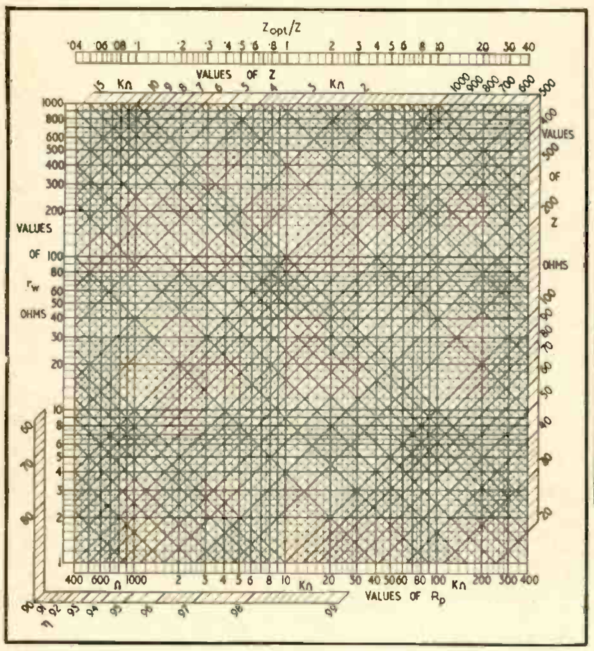

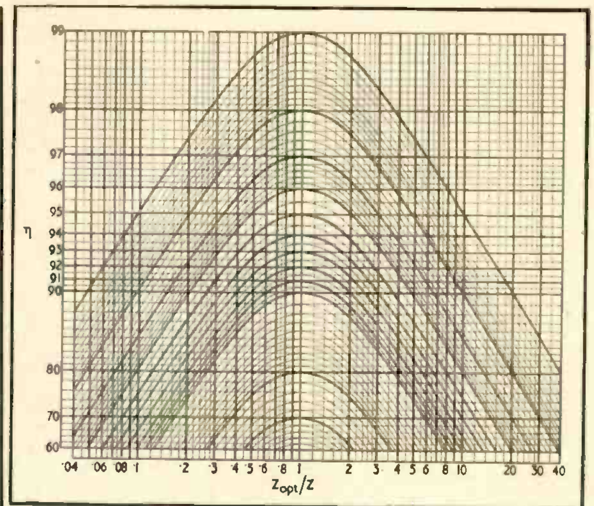

Fig. 5 (left). Chart to determine maximum efficiency, η, of a transformer, and the ratio of actual operating impedance to the impedance giving maximum efficiency. Fig. 6 (above). Chart applying the information from Fig. 5 to find the actual efficiency at the operating impedance.

r/R), allowing straight rulings to be used for frequency reference. Here the cramped spacing of the R/r scale toward the top gives an idea of the reduced effect on frequency when its value is large. The author has in preparation a book on the use of audio transformers, using charts with a different configuration again: a cut-out log scale can be applied to show the effect of variation in R, r, L, or C values more definitely; however, it was felt that this idea was not appropriate for a magazine article, so advantage is here taken of the construction yielding greater accuracy.

Adjustment of Values

Use of the 6 db/octave slope frequency as a reference does not mean that the frequency does not change at all as circuit values are altered, but that the change is easier to visualize or assess. If either $r=0$ or $R=\infty$ (the former never occurs and the latter seldom), variation of the other value would change the response in the way shown at *Fig.* 2 without changing the reference frequency. Values of R and r such that both contribute to the damping push the reference frequency upwards for h.f. circuits or downwards for l.f. circuits, to the degree indicated by the chart of *Fig.* 3. Variation of L or C will change the reference frequency in inverse proportion to the square root of their value, the chart of *Fig.* 3 again being used to calculate this.

Figure 4 shows how the chart of *Fig.* 1 may be used to visualize or estimate the effect of changing circuit values. The dotted lines on *Fig.* 1 represent a condition where L, C and either r or R are maintained constant, while the fourth quantity is varied. *Figure* 4 shows clearly the direction of movement along these dotted lines (or parallel with them) for increasing values of r and R respectively—decreasing values naturally produces movement in the opposite directions. Increasing L, or decreasing C, moves the reference point horizontally to the right, and increasing C, or decreasing L, moves it to the left.

In h.f. circuits, L is set by leakage inductance, but C can be increased by the addition of shunt capacitance across the high-impedance winding. In l.f. circuits, L may be variable by adjustment of the gap, and definitely does vary with signal level; C, the coupling capacitor, can be made any desired value.

No provision is made on the chart of *Fig.* 1 for $R=\infty$. In cases where a step-up transformer may be working without a secondary loading resistor, a high value may be assumed. It will be noted that the dotted lines representing variation of R, with other values constant, converge to become parallel with db rulings towards the top left hand corner of the chart. A high value of R will produce a reading in this area, where its precise value will not appreciably affect the db reading.

Efficiency

In transformer circuitry, insertion loss is more readily dealt with in terms of efficiency. Percentage efficiency can easily be converted to db insertion loss by log table, slide-rule, or one of the ready made conversion tables frequently published. Efficiency relates to power transfer—and not voltage transfer—so for example, an efficiency of 60 per cent is an insertion loss of 4 db, 90 per cent

[*Continued on page 74*]

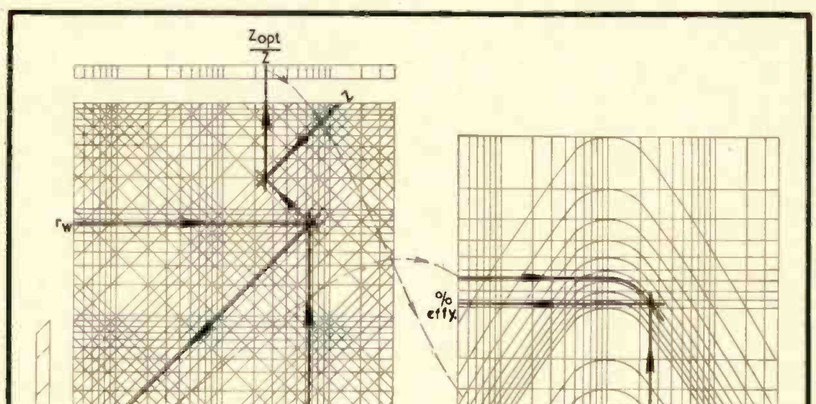

Fig. 7. Showing how the charts of Figs. 5 and 6 are used to find the efficiency of a transformer.

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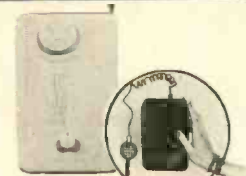

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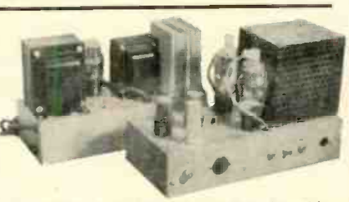

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EDWARD TATNALL CANBY*

SUSPENDED IN SPACE

WHAT ABOUT all this excitement over "binaural" recording, broadcasting, reproduction? It was only a year or so plus some, that we first heard of commercially available two-channel recording. via Magnecord. At that time the "binaural" Magnecorder was no musical instrument. On the first announcement, down in a small paragraph headed HOME, there was a brief mention of the enhancement of music by twin-channel recording; but the rest of the blurb was all about industrial uses, business conference work and the like. A lot seems to have happened since then!

"Binaural"—the quotes used advisedly—was the sensation of the recent Audio Fair and the pronouncements concerning its wonders were so joyous and jubilant that many went forth astounded. A few, I regret to say, went away convinced that the whole thing was publicity. It was surely that! The demonstrations were impressively staged and expensive, indeed a bit overpowering in their splendor. Wherever one went. there was "binaural," in an assortment of forms from earphones to batteries of trumpet speakers. Steam engines as big as a house snorted in and out deafeningly, shaking the rafters, ping-pong games were played twenty times life-size, musical comedies, symphonies, great organs peeled forth both loud and soft (but mostly loud). persuasive gentlemen announced, "binaurally," that they would now walk to the right . . . and then to the left . . . and few of the enthralled listeners were disturbed when for some reason they seemed merely to tromp up and down on the same spot, vaguely located near the ceiling. Soloists performed grandly out of the Right Speaker and the Left Speaker and only a handful were heard protesting that there were two of each, like double vision. "Binaural" music for you and me was everywhere proclaimed, but somehow the details were left a bit indefinite.

Quarter-Baked

Don't be misled—in the wrong direction—by all this "binaural" excitement, in case you have already discounted two-channel recording as one more publicity stunt. Maybe it won't be *that* good (nor that soon). But there is definitely something in it, behind the big-gun stuff now going on. The trouble with "binaural" music right now is that it is scarcely begun to be developed for practical use. A fine big chocolate cake, but somebody took it out of the oven quarter-baked. And the "binaural" programs at the Fair and via broadcast, in spite of some very excellent results, were in the large just that. Confusions, contradictions, false and faulty effects, happily swallowed up in the general excitement. Perhaps this is as it should be. We Americans usually launch our new projects in this high, wide, and handsome way. But somebody has got to untangle things sooner or later or the fine little "binaural" boomlet will collapse . . . well, like a quarter-baked chocolate cake.

This department did some speculating on this question a year ago (Æ, Dec. 1951), nor was that the first time "binaural" had been injected here, an earlier and very long-winded piece of mine (Æ, Jan. 1950) having had to do with the strange relations between binaural and monaural liveness, for recording and broadcast. Not a simple area at all, but the present "binaural" experiments have got us far more confused than ever before. I'll make only one major, fundamental point here, to illustrate the sort of straightening out that must be done before we know where we're going in this. Do we really mean binaural?

Binaural—Without Quotes

All recording and broadcasting is now done via one channel, with from one to *n* mikes and the same at the speaker end. Monaural. *Illusions* of reality can be accomplished extraordinarily well, given good microphoning and good listening acoustics in the reproduction. Nothing new about that. True binaural reproduction must, however, duplicate the two sounds heard by two ears—and, ideally, reproduce the effect of the original two eared listening. True binaural, effecting complete separation for each ear, is done only with earphones. The dual microphones are placed as nearly as possible to simulate a head with ears. No loudspeaker combination can possibly give a true binaural reproduction, since not only will the two channels overlap, both ears hearing both speakers, but there is a false sense of direction injected by the fact that the speakers themselves are directionally spotted—thanks to our hearing each of them with *both* ears. But do we need true, literal binaural? That is an interesting question.

Simple Substitution

There is an entirely different plan of action called stereophonic sound. This plan is simple. Forget about two ears, one channel for each. Instead, use two, or three, or numberless channels to put the sound sources spatially in the place where they ought to come from, right to the right and left to the left. With but two channels you can record a piano on one side of a stage and a singer on the other, each with his mike; then put speakers in place of the mikes, remove the performers and play the recording. You'll have them in their proper spots—because they *are* in their proper spots, by recorded proxy. Substitution. You can move the speakers about as if they were the performers themselves and have them "perform" in any old position you want. Nothing whatsoever to do with two-eared, binaural transmission.

For more than two performers, for a whole area of sound, more mike-record-speaker simultaneous channels are needed—theoretically an infinite number. But, as was shown 'way back in 1934 by Stokowski *et al*, three channels—one to each side and one for the middle—will give an excellent and close approximation of full substitution coverage. You'll find a similar system in use now for the fabled "Cinerama". It is *not* binaural. No earphones! (Nor is "Cinerama's" picture a stereo picture—one picture for each eye. Its several pictures are spread out exactly as is the sound, the simultaneous projections overlapping at the edges as do the sounds from the simultaneous recording channels.)

20 Feet

Now, my chillun', for the $64 question. Just which of these systems are we trying to use, in our so-called "binaural" recording?

It should be apparent to you that we have been dabbling heavily in both, mixed together. We have had something of everything.

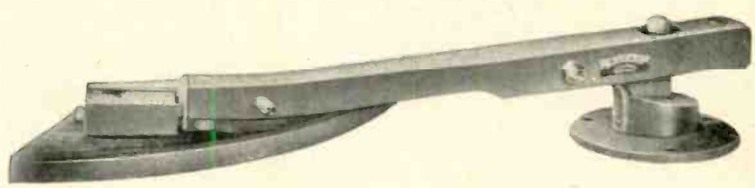
* *780 Greenwich St., New York 14, N. Y.*

In music,
listening
quality
is everything . . . .

"My wife and I drove some 400 miles to New York, just to witness that concert of yours at the Audio Fair. For over an hour we waited in line, but it was worth it. No amount of advertising could have made me believe such a thing possible. Closing our eyes, we could not tell which was playing—the live artists or the AUDAX CHROMATIC. Thank you for this thrilling experience. It will stay a memorable event in my musical life. I am going to . . . ." *(from a letter)*

The revolutionary new records are so true to the original that almost any pick-up is bound to give some results . . . but—it takes a reproducer of the highest order, one sensitized to the nth degree . . . a CHROMATIC POLYPHASE—to bring out every subtle shading, every nuance so essential to the real music of which these discs are capable. . . . Therefore . . . SEE and HEAR CHROMATIC and —YOU be the judge.

- Audax costs no more than ordinary pickups
- One single magnetic pickup plays all home records
- Only AUDAX has individually replaceable Sapphire or Diamond styli.
- Response 10 to over 15,000 cps.
- Needle-talk practically nil*
- Near-infinite compliance
- Phenomenal tracking



Available with the new compass-pivoted Audax arms and to fit record changers. Audax arms are the simplest and most efficient yet devised

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AUDAK COMPANY
500 Fifth Avenue New York 36

Creators of Fine Audio-Electronic apparatus for over 25 years.

"The Standard by Which Others Are Judged and Valued"

We have had mikes at 20 or more feet separation—fed into earphones, as though for a binaural effect. Do we intend, thus, to make a "giant" head for ourselves, putting our two ears thus far apart? We do just that, if we play the result through earphones, filling each ear totally with the sound from one mike alone. The two sounds can then only be fused inside the head. The earphones obviously are not located at the place where the sounds are supposed to come from; we'll have to do the locating mentally, via interaction of the two ears. That's binaural, pure and simple. No substitution.

But play the same recording, with 20 foot mike spacing, into two speakers 20 feet apart, and you have something else again. Very different. The speakers are actually in the positions of the original sound, more or less. They act as substitutes, proxies on the spot—and this is the substitution system again. Stereophonic.

We've also had much two-channel recording (or broadcast) with mikes close together, as on a head more or less. Say between 6 and 20 inches. A slightly swelled head, but a head, nearly enough. What happens when such a recording is played through loudspeakers spaced apart—say 20 feet? Interesting. The close-to mikes can give no substitution effect, on the spot. Too close. Whatever effect there is must be binaural—the interacting of the two sounds inside your head, to produce a *seeming* direction—if there is any direction-effect.

$10,000 question

Is it possible, then, to get a *binaural* effect in any degree with two speakers—i.e. an effect that is independent of the actual position of the two speakers themselves, as you look at them? To put it concretely, can you record a single person's voice, say, two-channel via two closely spaced mikes, then play him back through two speakers widely spaced—say 20 feet apart—*and have just one of him appear, midway in space between the two speakers?* This, rather than two of him, one coming from each speaker. That, my friends, is the $10,000 question. Direction is only one aspect of binaural realism, to be sure. There's lots more to it. But if you can achieve that, you'll have the rest in the same measure. And music in the home will be impressively benefited.

Let's pause here, for a good breather. I'll say only that once—just once, so far—I have had definite proof that this is possible. I heard my own voice suspended in space between two speakers. Just one of me. I'm going to do my best to find out how I did it!

RECORDS

Opry and Drama

(There are such floods of standard repertory opera now appearing on LP that this department takes a big breath and swims dizzily towards the quieter reaches—herewith a few unusual works of great interest.)

Mozart, Zaide. Paris Philharmonic; Mattiwilda Dobbs, Hughes Cuénod, Bernard Demigny, etc., Leibowitz.
Polymusic PR 901/2

Old Mozart fans will gulp at this—for it is only half an opera, unfinished, and was never performed; few music lovers even know of its existance. Not a minor opera, either—this would have been one of the major works; but for reasons probably commercial, Mozart casually dropped it in the midst and went on to another which was commissioned and would, he knew, be performed. Zaide was, however, re-done in the later work, Die Entführung, one of the gayest, silliest, most brilliant pieces of Turkish slapstick you can imagine. (London has it on ffrr.) This one was to have a similar Turkish harem sort of plot. Only the arias for two acts and some wonderful "Melodram"—spoken text with musical accompaniment and interludes in a style suggesting today's radio technique—were completed.

A good performance, in spite of the polyglot cast, a French orchestra and conductor, a Swiss hero (Cuénod) and so on; the playing is a bit rough and a trace hard, but the essential Mozart gets over beautifully, especially through the superb singing of the leading lady named Dobbs. If you enjoy this sort of Mozart, here is a wealth of new material of the first rank.

Purcell, Dido and Aeneas. Flagstad, Schwarzkopf, Mermaid Theatre Company, Jones. **HMV LHMV 1007**

The first, and perhaps greatest English opera, here at last complete and done in the original shape without fancy "arranging"—but the feature of this HMV LP is Flagstad, who has never sung more beautifully, nor been recorded more gracefully. If you don't know Henry Purcell, the first time through may leave you with a feeling of strangeness—but in no time the wry, penetrating strength of this composer will get you, the silly text ("Our plot has took, the Queen's forsook!") will reveal the tragic emotions beneath it. Elizabeth Schwarzkopf is also superb—both ladies sing and speak perfect English, in the purest English manner; the direction is rhythmical and lively; the orchestra, harpsichord, and chorus all tops. Recording is unusually fine, with but one drawback: the LP sides are too long, with considerable distortion in the inner grooves, as heard wide-range.

Mozart, Cosi fan tutte. Steber, Thebom, Peters, Tucker, Metropolitan Opera Orch. and Chorus, Fritz Stiedry. **Columbia SL 122 (3)**

You'll have to buy three LP's here, to experience some of the finest technical opera recording yet done. Whatever the method (and artificially added liveness is a good bet), the effect of depth, perspective, "width" in this sound is no less than fabulous. Shows what extraordinary illusions can be created in the monaural medium, given the right mike setting. The orchestra seems huge, yet is not, and each instrument is ultra-clear, the drums full and sharp, the strings close and stringy (close-up mikes?). The voices are recorded quite close, a convention which, if quite unfaithful to the original opera-house sound (the voices at stage distance), is clearly the best way to get opera over on records.

The opera itself? First Mozart to be done complete here, it is in English—the frothy, silly but ultra-clever little farce translated into language that—thank Heaven—not only makes reasonable sense but, more important, is beautifully fitted to the music. Sounds natural—which is a feat in opera-in-English! Very complete notes and comment are a tremendous help in the listening. This is the way to hear opera! When you know it cold, on records—then go to the opera house and see it in the flesh, with enormous appreciation. Lively performance under Stiedry, the slighty Americanized atmosphere following naturally from the American text. You can understand every word.

Offenbach, La Belle Helène. Linda Felder, André Dran, Paris Philharmonic Orch. and Chorus, Leibowitz. **Renaissance SX 206 (2)**

Know French? If you understand three words of it, this opera—operetta, I should say—will knock you over with a feather, to mix metaphors. This is the ultra-French version of Gilbert and Sullivan, mixed up with a considerable dash of what might be U. S. musical comedy. Preposterous plot involving the great Greeks—who turn out to be side-splitting imbeciles, canny plotters, the story making about as much sense as G & S, moving at least ten times as fast in the most incredibly clear and lightning-speed French, both spoken and sung. It takes the French to put on a whirlwind show like this, screamingly funny, yet with a technique that is breath-taking in its virtuosity. You'll recognize a lot of the music and, with the complete printed text, will be delighted at the book. Just repeat "L'homme a la pomme" at a speed of 100 times a minute and you'll get the feel of the show. That's one of the choruses. Terrific.

NOVELTIES

Mozart, Piano music (Sonata, K. 533; Rondo, K. 494; Adagio, K. 540; Fantasy and Fugue in C, K. 394; Rondo, K. 485.) Paul Badura-Skoda.
A. On a modern piano. **Westminster WL 5154**
B. On a piano of 1785. **Westminster WL 5153**

Mozart, Piano Sonata in B flat, K. 570; Suite, K. 399; Fantasy and Fugue in C, K. 394. Ralph Kirkpatrick, Challis 18th century type piano. **Bartok BRS 912**

Here is a challenging musical and intellectual experience for the curious minded. Mozart wrote, in the late 1700's. when the piano was scarcely begun on its modern development—yet he obviously found it good for his purposes, as did his listeners. Is Mozart improved when played on our perfected modern piano? Is the piano itself better now?

Tricky questions. Technically the modern piano is of course "improved." Louder, smoother, more sure in its action, bigger, and so on. But mechanical improvements often change the character of an instrument. A skilled composer writes for his instrument, as is, and takes advantage of its qualities, even the "bad" ones, for positive musical ends. The plain fact is, as these records will persuade you, that Mozart sounds extremely well on a Mozart-period piano—perhaps a good deal better than on the modern instrument. The peculiar soft, twangy tone qualities that, for later music, might seem ugly, are for this music rather perfectly suitable. The effect, paradoxically, is richer, more emotional than on the big new piano —perhaps you will now discover why Mozart so often sounds thin and unimportant on our instrument.

Badura-Skoda, a young player who is one of the finest Mozart pianists we have, gives us a splendid working comparison on the two pianos, my only reservation being technical: from what I know of the instrument, I suggest that the old piano's bass is somewhat unnaturally heavy and big here. This is a rebuilt Viennese piano dating from Mozart's day. Ralph Kirkpatrick, harpsichordist and authority on the older music, gives a good performance too, of a drier, less fluent sort (influenced, I'd say, by the harpsichord style of playing). His piano is a newly built one, a compromise reconstruction taken from a number of different piano types of around 1800 "averaged" together. The Bartok recording is superb, giving a better idea of the instrument than the Westminster.

What are we coming to—when, after a century of constant technical improvement, we now go back and deliberately build a Mozart piano, reversing the entire piano development! Same thing for the so-called "Baroque" organ, now so popular; it is no more than a revival of 17th and 18th century organ mechanisms, on which to play the music of that time, for better effect. Makes one think a bit.

Schubert, Symphony #7 in E major, arr. Weingartner. Vienna State Opera Orch., Litschauer. **Vanguard VRS 427**

It is not often—or it was not often—that a major work of a supremely great composer appears on the scene for the first time. Here, for most of us, is such a work. The 7th symphony was left by Schubert in sketch form, "complete," but in a sort of shorthand. (How many of us leave unfinished projects about in this fashion, waiting for the convenient time to do a bit more hard work. . .) It was not heard until Weingartner, the conductor, made a playable version of it in 1934 for orchestra—here recorded.

This symphony (and we hear substantially what Schubert intended) comes between the early six, of which several are now very popular, and the Unfinished, of endless fame. Between that and the "Great C Major", almost as popular, came that other mystery-work, the "Gastein" symphony, which disappeared without a trace but is presumed to be the same music as the Grand Duo for two pianos opus 140. It was recently recorded, on this assumption, in an orchestral arrangement approximating what Schubert intended. (Vanguard VRS 417). What a human and fallible man this Schubert was—his musical affairs in a constant state of semi-confusion, his fantastic musical ability pouring out new music faster than he could organize it into properly finished products!

Both the "Gastein" and this E Major work are thoroughly big pieces; both are full of the things we know better in the familiar Schubert. If you have been captivated by the Great C Major (wrongly called #7 or #9, though it is actually #10)—or by any of the fabulous Schubert works for smaller instrumental combinations—you'll not go wrong on this one. Sympathetically, rather mellowly played by the Vinennese orchestra.

Bird Songs of Dooryard, Field and Forest. Recorded by Jerry and Norma Stillwell. 12" LP (Ficker Recording Svce., Old Greenwich, Conn.)

The first LP bird record (though Folkways Records has a South American Rain Forest recording of mingled sounds), this one follows roughly the pattern of the famous Cornell recordings issued in 78 r.p.m. form—spoken comment introducing the birds, pointing out features of the songs, etc. The recording quality of most of these is not as good as that in the Cornell records; there is much buzzing distortion, neither the highs nor the "lows" are as clear, and the birds seem often unnaturally close. (Both recordings use the parabolic mike extensively.) But the Stillwells have done good things, in comparing a series of different individuals of one species, in playing some songs half-speed, "slow motion," in the wealth of specific material on how to distinguish one bird from another. The long-play medium allows for greater leisure, longer excerpts. An important documentary disc as well as a most enjoyable production.

Anna Russell Sings? (Advice on Song Selections for Concert Singers.)
Columbia ML 4594

This dizzy dame, recorded at a "recital", is a take-off genius, making delightful and very musical fun of any number of styles and manners and schools of singing. The more you know of music the more pointed her satire—but the stuff is wonderful (in smallish doses) for anybody at all. Fine sense of presence, audience participation, belly laughs—on the record—until your own sides ache. Her English accent is lahvly!

PROTECTION

(*from page 36*)

will be of sufficient value to merit a reasonable charge therefor."

No response was made to this letter but later, when the suggested idea was put into operation, the Indiana court said, holding the originator entitled to recover,

"While we recognize that an abstract idea as such may not be the subject of a property right yet when it takes upon itself the concrete form which we find in the instant case, it is our opinion that it becomes a property right subject to sale.

"Of course, it must be something novel and new; in other words, one cannot claim any right in the multiplication table."

References

Millar v. Taylor, 98 Eng. Reprint 201, 234
Pidot v. Zenith Radio Corp., 31 N.E. 2d 385, Illinois
Bristol v. Equitable Life Assur. Soc., 30 N.E. 506
Haskins v. Ryan, 64 Atl. 436; 78 Atl. 566, New Jersey
Lueddecke v. Chevrolet Motor Co., 70 Fed. 2d 345
Masline v. N. Y., N. H. & H. R. Co., 112 Atl. 639, Connecticut
Liggett & Myers Tobacco Co. v. Meyer, 194 N.E. 206, Indiana

PATENTS

(*from page 6*)

curve 2; too low, like curve 3. Though the patentee does not say so, it seems that a safe setting to take care of tape variations would be a bit high but not so high that dynamic range is limited by nonuse of too much of the straight portion of the *B-H* curve of *Fig.* 1. Connell claims that his system gives high-quality results, without any of the usual bias-caused deterioration of the high frequencies.

Wide-range Probe

Today audio amplifier circuits have reached such a state of perfection that often the usual measuring instruments are not good enough to cope with them. For gain measurements or oscilloscope observation in early stages at high impedance, for example, the measuring instrument leads or probe must have negligible resistance and capacitance, which is far from true for most.

Edward W. Yetter has designed an instrument input stage which is simple and small enough to be built into a probe of reasonable size but has extraordinarily good characteristics. The patent covering it is No. 2,601,485, and it is assigned to Sun Oil Co. The probe has an input capacitance ranging around .01 μμf, input resistance of 1,000 megohms or so. It is linear for all input voltages from zero to around 100 volts and up to 1 megacycle. Its output is at low (cathode) impedance and can be fed to any sort of measuring equipment at all.

The circuit is diagrammed in *Fig.* 6. there are two tubes, a 954 acorn-type pentode and a 12AU7 miniature duo-triode. Probably the only component which need be mounted in the probe is the 954, with a short probe wire soldered or clipped to the grid pin.

The 954 is operated as a cathode follower. While the permissible range of input voltage of a cathode follower is greater than with plate output, it is still not particularly high when linearity is to be maintained. This is because of variations of potential difference between cathode and screen. The purpose of the 12AU7 is to maintain the screen-cathode voltage substantially constant so that the voltage range of the probe will be very wide.

Output is taken from the junction of R_1 and R_2 for the instrument being used. The value of R_3 plus R_4 is adjusted for linear operation of the first 12AU7 triode. Its grid is excited from the tap on potentiometer R_1. With the two cathodes of the 12AU7 tied together, the second section is operated almost as a grounded-grid amplifier and the combination is simply a phase inverter. The cathode-plate path of the second triode acts as the grounded leg of a voltage divider, the upper half of which is R_7–R_8; screen voltage from the pentode is taken from this divider.

When the probe voltage goes positive, plate current in the 954 increases. As is usual in cathode followers, the cathode also goes positive, and some of the positive voltage is coupled to the grid of the first triode.

The first triode is also a cathode follower and its own cathode goes positive, as does that of the second triode ,which is directly tied to it. The positive triode cathode voltage appears across R_5–R_6 to ground, which means that the grid of the second triode is somewhat negative with respect to its cathode. For that reason, plate current in the second triode falls; the tube is effectively a higher resistance and its plate goes positive. The positive-going voltage at the plate of the second triode is the screen voltage for the 954. Thus, while the probe signal caused the 954 screen to go negative because of increased current flow, the second triode compensates for this by bringing it positive again. The amplification added by the triode to the compensating signal actually produces a positiveness which is more than proportional to the original change in 954 screen voltage. In this way, instead of going negative with increasingly positive 954 grid and cathode voltage, the 954 screen actually goes more positive than before to keep in step with its own cathode voltage. This maintains a fairly constant screen-cathode potential, preventing distortion for probe signals as high as 100 volts.

The four potentiometers need be adjusted only once, then locked. The idea is to adjust them for linear operation at the highest possible input voltage.

The additional features of wideband operation and low input capacitance are, of course, inherent in cathode followers, but they are bettered by the screen compensation as well.

A copy of any U. S. patent may be had on request from The Commissioner of Patents, Washington, D. C., for 25¢.

LOUDSPEAKER

(*from page 34*)

brought together you have great music.

Now it is conceivably possible that all this complicated ritual might be scientifically analyzable, but I do not think it has ever been done. The result is determined by long empirical experiment both in making the instrument and in playing it, and the excellence of the result is determined by aural testing in the presence of musical experts. The Amati family, for example, found by repeated trials that a certain type of wood, cut and shaped in a certain way, assembled in a particular manner and varnished with a lacquer of a certain composition, produced violins that were esteemed by musicians, and they had not the foggiest notion of the mathematical principles underlying their work, nor—may we hazard a guess?—has anyone else. Their instruments had a certain "tone color" which was greatly admired by violinists and musically conscious listeners, and how they got that tone color was their family secret, found out by empirical experiment.

When one comes to consider the design of a loudspeaker a similar state of affairs exists, but the aim is different. A speaker is not a musical instrument, it is a reproducer of musical instruments, and therefore it must have no tone color at all, otherwise its reproduction of the original music will be false. There is no way of scientifically determining beforehand how to achieve this desirable property, and when the design has been completed there is no way of finding out if the design is successful except by listening to it. And that is where the trouble starts. That is why so many musicians do not like that type of reproduction commonly called "high-fidelity." They seem to prefer the sort of distortion that one gets with "middle-fidelity" rather than the distortion one gets with high-fidelity. It is useless to point out that "high-fidelity" means freedom from distortion, for freedom from

distortion in that sense only implies that scientific measurement does not reveal distortion. The non-musical scientist may "prove" that there is no distortion, but the musician will not be impressed with such a "proof." He will merely point out that it doesn't sound like the real thing, and that is enough for him.

Checking the Results

With this in mind we can now consider my presupposition that the speaker diaphragm is something flexible which flaps about in the air to produce a complexity of sounds. I have shown that mathematical analysis of this is impossible, but scientific experiment has been able to lay down certain desiderata. It has been shown, for example, that a restricted frequency response will not permit of satisfactory reproduction of music—there is lack of bass and wooliness owing to suppression of the higher harmonics of the instruments being "reproduced"; speech, also, is "plummy." Accordingly, for what is called high-fidelity reproduction, a frequency range of about 50 to 15,000 cps is postulated. But in addition to this there must be good reproduction of transient noises otherwise the reproduction will lack what is usually termed "attack." Apart from wide frequency response this requirement postulates a very low degree of inertia in the reproducing system, otherwise it will not be able to deal with sudden impulses without delay, or return to the normal sufficiently quickly to deal with the next impulse. If we assume that the design of the amplifier has been completed satisfactorily (and the design of this can be done mathematiaclly, and tested scientifically, because it is designed to handle electrical currents which can be measured for amplitude, frequency, phase, and transient distortion) we are left with the problem of designing a physical body which has to produce audible sounds which can only be partially measured.

If the speaker cannot be designed by scientific methods it can only be done by trial and error, and this requires a knowledge of what is required, how it can be done, and how it can be checked. Leaving out the magnet system and the voice-coil assembly, we are left with the problem of designing a diaphragm which will flap about in such a way that the sum total of its flapping is the reproduction of all musical sounds within the prescribed limits without audible distortion. How this can be done does not come within the scope of this article, but how it can be checked has already been mentioned—by listening to it. You may object that it is not fair to condemn a certain technique without offering a replacement technique, and provided the terms of reference of the argument had not been laid down the objection would be valid.

Terms of reference were, however, laid down. They were: an injuiry into the contention that the perfect diaphragm should be an infinitely rigid disc of no mass is absurd and also my assertion that if such a diaphragm could be produced it could only reproduce a sine-wave. From this inquiry emerged an argument which suggests that the mathematical design of a loudspeaker is impossible, and if this is so, then the thing can only be done by trial and error guided by practical experience. Finally, it can only be assessed by subjective artistic standards. That is the result of adopting my absolute presupposition, but as the two opposing presuppositions are by their very nature articles of belief, since there is no *a priori* knowledge to which you can refer, your acceptance of one or the other is a matter for your taste and logical conscience. But your choice will determine the final result.

In selecting a loudspeaker to entertain and instruct you you can, on the one hand, go to a great deal of trouble in finding out all about it in a technical way, having previously armed yourself with the technical specification of what is required electrically to create undistorted reproduction. You will demand response curves, figures for electro-acoustic efficiency, intermodulation distortion, and anything else that occurs to you, and when you have made your choice you will erect the reproducer and say "That is the last word on high-fidelity reproduction." You will, in fact have joined the ranks of what some people call "audiophiles," and you will be hurt and annoyed when some ordinary individual comes along, listens, and says "I think it is lousy." You will dismiss him as a man who knows nothing about musical reproduction.

On the other hand you can say "I do not believe that the sound of an orchestra can be defined by a mathematical equation. When I go to a concert I hear things which please me or annoy me, but whatever I hear is the result of people consciously making music. What I want in my home is as close an approach as possible to the sensation created in me when I go to a concert." And the only way you can assess the qualities of any equipment offered to you is to listen to it, taking very great care that you are comparing its performance with original music and not with other equipment. If this is how you select your equipment, then the very last place to go to is one of those "audio test rooms" wherein a dealer will let you hear twenty different speakers working on twenty different amplifiers. In such a place you will certainly be able to compare one equipment with another, but your ears will become so "conditioned" to the sound of an electronic device that you will forget what happens in real life. If you do not believe me, go and listen to all the equipment you can hear in any big city, try and carry it in your mind. And then go to a concert. The shock is profound. You will, in fact, realize, as others have done before you, that "high-fidelity" is getting farther and farther away from realism. Without the audio engineer you cannot have the foundations of realism, but without the musician you will remain an audiophile.

Equipment Report

SONOCRAFT FM TUNER

Compact new FM tuner offers advantages for installation in small cabinets, yet provides satisfactory reception even in fringe areas.

For high-fidelity radio reception, it is generally conceded that FM provides better quality and less distortion than AM, and listeners located in cities where FM stations carry the programs of local or network stations simultaneously usually prefer to use only one type of receiver —FM. Actually, a good AM receiver is capable of high-quality reception, particularly when the receiver is of the t.r.f. type, but while such receivers are satisfactory in areas close to the transmitters, they are not nearly so desirable at long distances, because of atmospheric noise. Therefore, the listener desiring good quality will usually turn to FM, even if he may be in what is commonly termed a "fringe" area.

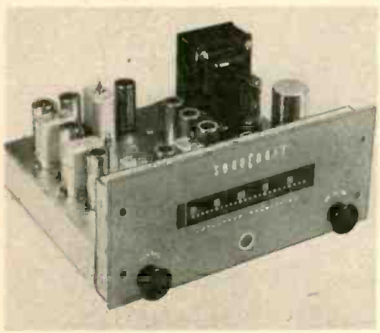


Fig. 1. External appearance of the tuner, which has a panel only 10-in. wide and 4¼ in. high.

FM reception in outlying areas offers a problem to many listeners, however, due to lack of sufficient signal strength. The Sonocraft FM tuner, however, has excellent sensitivity, and under test conditions at a location 40 miles from New York performed very satisfactorily.

The tuner is of conventional design, except for the r.f. and oscillator circuits, which are slug-tuned. It employs a ganged assembly of cores which move in and out of the coils, and the construction is such that trimming may be accomplished at both ends of the tuning range. The i.f. amplifier consists of three stages, followed by two limiter stages and a discriminator. The 6AG5 r.f. stage provides plenty of signal sensitivity, the figure claimed by the manufacturer being 5 microvolts for 20-db quieting. The bandwidth at 6 db down is 200 kc, with sufficient selectivity to reduce adjacent channels by 50 to 60 db.

The tuner is equipped with an effective a.f.c. circuit, which may be switched on or off from a front panel control. With the a.f.c. switched off, the drift is excessive for the first ten minutes of operation, but thereafter the drift is negligible over a period of hours.

One feature of the tuner is the indicator, which is simple and of clever design. A 6J6 serves as the indicator driver, being connected as a v.t. voltmeter tube. The indicator tube is an NE-51, connected between the plates of the 6J6. One plate of the NE-51 glows when the set is tuned on one side of the correct frequency, and the other glows when the set is tuned to the other side. When the set is correctly tuned, neither plate glows—from which the tuning indicator derives its name, "Blackout."

Because of its compactness, this tuner is adjudged to make a satisfactory addition to a home music system in which only FM is desired and which must be housed in small cabinets. It is self powered, and is not equipped with a volume control, since the tuner output is usually fed directly into the control amplifier.

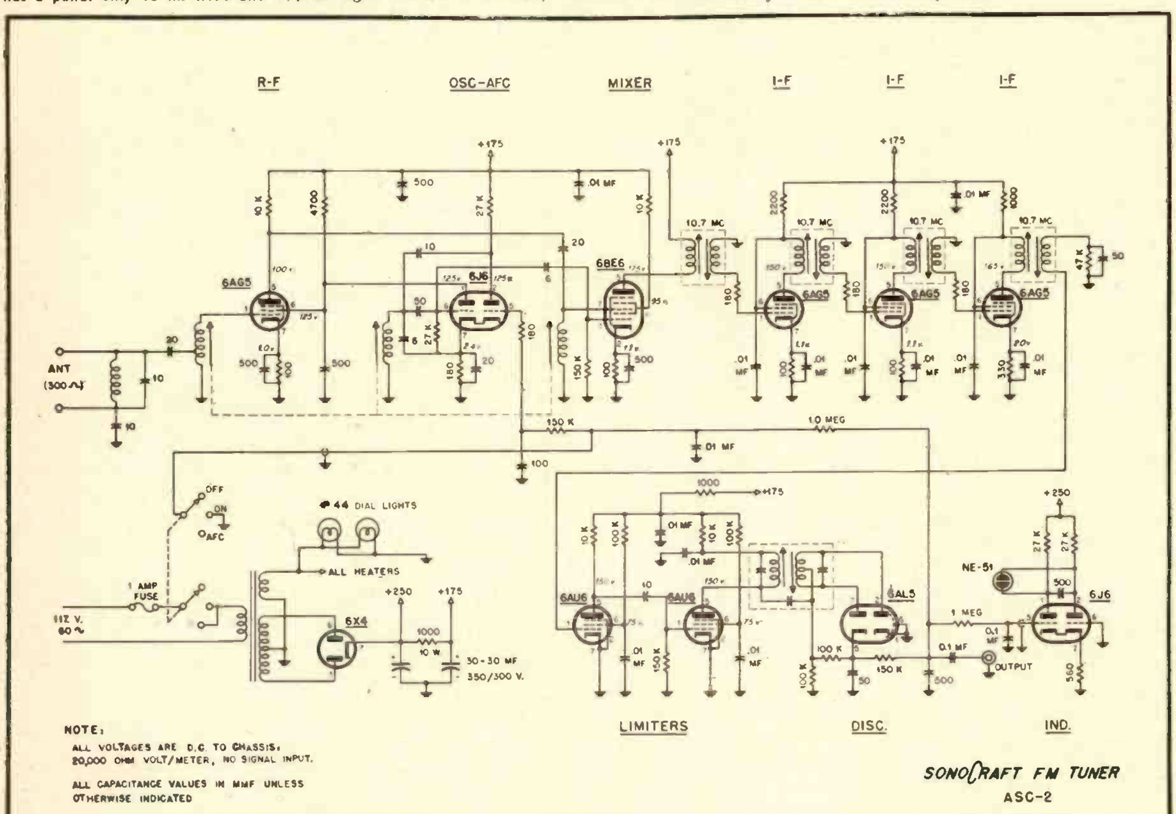


Fig. 2. Complete schematic of the Sonocraft FM Tuner, with operating voltages.

BINAURAL OR STEREOPHONIC?

[*from page 24*]

Something should be said about suitable tape recording equipment. *Figure* 4 shows a typical two-channel portable recorder. One case contains the tape transport mechanism, another contains two separate record and simultaneous playback amplifiers, one set for each of the two tracks. The third contains the power supplies for the amplifiers. The erase/bias oscillator, mounted in one amplifier, supplies bias directly to one channel and through a buffer amplifier to the other, so that the same bias frequency is used in each recording head, thus eliminating the beat frequency which would result from using two separate oscillators of approximately the same frequency. *Figures* 5 and 6 show the head assembly with a full track erase head, two side-by-side record heads which record two simultaneous tracks, running parallel along the length of the tape, and two appropriate pickup heads. Tape motion is from left to right across the heads. Thus the tape is first erased, then recorded upon, and then monitored a fraction of a second later over the playback system. The record heads are well shielded between the two and have 50 db attenuation on cross-talk between channels. The gaps of both the record and playback structures are critically aligned in parallel so that the time error between the two separate tracks is less than .00003 sec. at 15 in./sec. tape speed, or .00006 sec. at 7½ in./sec. This has an interesting sidelight. Tapes recorded stereophonically (or binaurally) on this type of inline gapped head may be played back on a standard full-track reproducer in an entirely satisfactory one-channel manner. This makes it possible to produce pre-recorded stereophonic tapes which may be played on either a stereophonic or a standard playback machine.

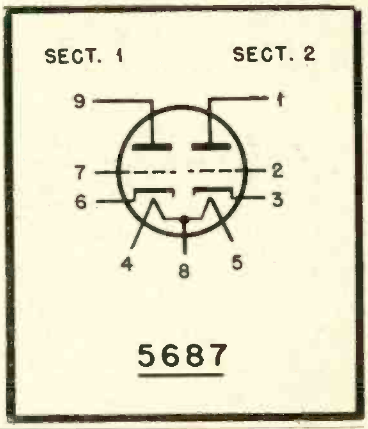
It will be noted that binaural headphone listening to orchestral music, for example, appears to give a much greater sensation of a completely "new" way of listening, and that loudspeaker stereophonic listening (correctly done) is not as astonishing. Headphone binaural listening is best for many special investigative problems. But speaker reproduction should sound more natural than the headphone method because we are used to being in a room-acoustic environment. In fact, it is for this very naturalness that we search.

The author is indebted to his early teachers in acoustics, his many customers, his associates, and several midwest university musical organizations for background and experimental set-ups which have led to the conclusions set forth above. Some of the conclusions reached might be somewhat controversial. Much three-dimensional acoustics still is. As one acoustican has said: "In no branch of science is the theory so simple and the measurement so difficult."

5687 Basing

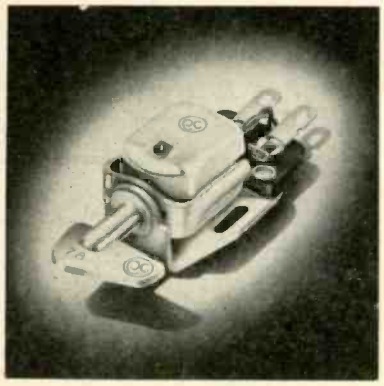
The Tung-Sol 5687—used in the Kiebert version of the "Williamson Amplifier Up-To-Date" described in the August issue—is *similar* to the 12AU7, but is not directly replaceable for the latter tube. The principal difference of importance is the basing, with that for the 5687 being shown at the right. Note that the heater center tap is on terminal 8, rather than on terminal 9 as in the 12AU7. Note also that the connections to Section 1 of the tube are entirely different.

While the 5687 has certain advantages over the 12AU7, the user is cautioned not to attempt to replace directly without change in socket connections.



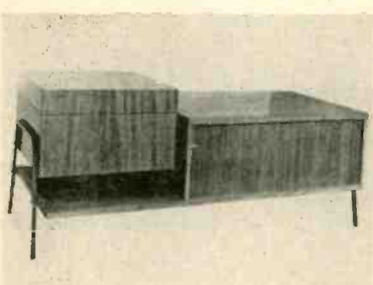
NEW PRODUCTS

• **New Turnover Cartridge.** Pickering's latest contribution to the technique of re-creating recorded music is their new Model 260 Turn-Over Pickup. The factors which have earned the present high reputation for Pickering Audio Components are inherent in the development, engineering,



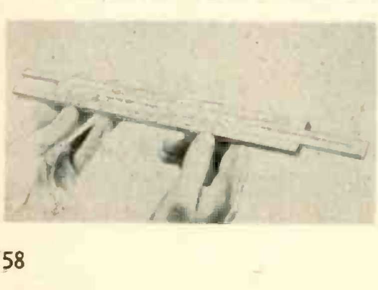
and manufacture of this new model, which has an output of 30 millivolts at a stylus velocity of 10 cm. per sec., and which mounts easily in any type of arm. Detailed literature may be obtained by writing Pickering & Company, Inc., Oceanside, N. Y.

• **Radio-Phonograph Cabinet.** A compact, convenient, and well-made piece of furniture is currently being offered by Jeff Markell Associates, 108 W. 14th St., New York 11, N. Y. Featuring a modern design in 12 finish colors on birch, mahogany, korina, walnut, and oak, these cabinets provide space for changer, tuner, and am-



plifier, and are of modern design, using half-inch square wrought iron legs. In addition to its duty as a housing for audio system components, the RC-118 also serves as an attractive cocktail table. The cabinets are available from dealers and jobbers.

• **TV Signal Range Calculator.** The new Pioneer calculator for determining the signal range of TV stations quickly shows the approximate Grade A, Grade B, and "Principal City" coverage for all VHF and UHF television channels. Resembling a slide rule in appearance, the instrument readily shows approximate field strength

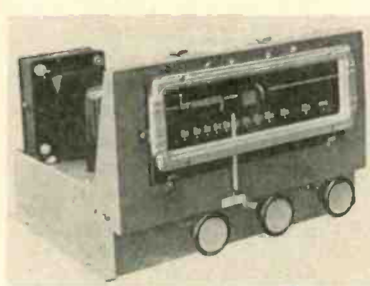


in microvolts-per-meter for distances up to 100 miles from a TV transmitter. Coverage radius is read directly, and with one rule setting, for stations operating with ERP from 10 to 1000 kw, and for antenna heights up to 3000 feet. Pioneer Electronic Supply Co., 2115 Prospect Ave., Cleveland 15, Ohio.

• **45-RPM Recording Disc.** Said to be the first 45-rpm recording blank to be marketed commercially, the new Soundcraft "45" is made with a perforated center which permits use of the disc with all types of recorders and players. With the center section in place, the disc fits the

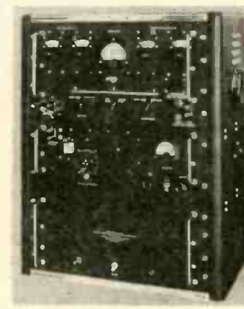
conventional recorder spindle. When the section is punched out, the disc is suited for use with standard 45-rpm turntables. The "45" is now in production and available for delivery. Reeves Soundcraft Corporation, 10 E. 52nd St., New York 22, N. Y.

• **FM Tuner.** An all-triode r.f. section and dual cascade limiters are among the features of the new Browning Model RV-31 FM tuner. Input signals of 3 microvolts will afford 20 db quieting; a.f.c. locks the local oscillator into correct tuning and may be switched off if desired. Selector



switch permits FM, TV, Phono, or recorder playback to be fed through the tuner volume control to the basic amplifier when desired. Extremely well-adapted to custom installation, the RV-31 is only 11 in. wide, 9⅝ in. deep, and 6¼ in. high. Full 15-kc audio output is fed through a cathode-follower output stage, thus permitting long cable runs without affecting high-frequency response. Power supply is self-contained. Browning Laboratories, Inc., Winchester, Mass.

• **Noise and Distortion Analyzer.** Design engineers in many phases of the electronics industry will welcome the Model ND-110 analyzer recently introduced by Empire Devices, Inc., 38-25 Bell Blvd., Bayside, N. Y. In essence the ND-110 is a harmonic wave analyzer capable of measuring the mean power of complex signals (broad band), and the distribution of this power within the ultrasonic frequency spectrum (narrow band). It may also be used to determine signal-to-noise ratio, distortion and intermodulation characteristics of telephone transmission equipment and various ultrasonic

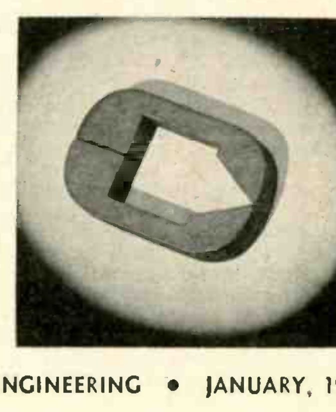


devices. Frequency range is 4 to 110 kc. The local oscillator is housed in a thermostatically controlled oven for maximum stability.

• **High-Quality Cabinet Speakers.** An exponential horn, compactly folded within an enclosure, assures excellent audio quality in a new series of speaker assemblies recently announced by White Sound, Inc., Suite 1105, 105 W. Madison St., Chicago, Ill. Available in four sizes, the new speak-

ers are designed on the principle that there must be a proper relationship between driver and housing if ideal performance is to be achieved. As a result, all cabinets are supplied complete with a speaker unit which affords optimum audio quality. Folded horns are within 2 per cent of a true exponential formula. Corner installation is not necessary.

• **Ferrite Recording Head.** Manufacturers of tape recorders will find interest in a new recording head which makes use of



a Ferroxcube non-metallic ferromagnetic core. A new material, designated Type 1-20-1, has been developed especially for use with magnetic recording equipment. It is very homogeneous and is more nearly free from voids and cracks than most commercially available ferrites. Technical information will be supplied upon letterhead request to the Application Engineering Department, Ferroxcube Corporation of America, 241 Marshall St., North Adams, Mass.

• **Junction Transistors.** The immediate availability of two PNP Germanium Junction Transistors types CK721 and CK722 has just been announced by the Receiving Tube Division of Raytheon Manufacturing Company, 55 Chapel St., Newton 58, Mass.

Both of these types of transistors have noise factors averaging 22 db at 1000 cps, with the type CK721 showing an average power gain of 38 db and type CK722 showing an average gain of 30 db. The units require a volume of 0.03 cu. in., and the leads may be soldered or welded into the circuit or cut for insertion into standard subminiature sockets. Both types are described in data sheets now available from Raytheon's Technical Information Service, at the above address.

• **LP Test Recording.** There is hardly any turntable or record-changer test which cannot be conducted with a new disc recently produced by The Dubbings Co., 41-10 45th St., Long Island City 4, N. Y. The record consists of two identical sides,

each of which contains 20 test bands. In addition to frequency checks from 30 to 12,000 cps, test bands are available for determining amount of rumble, correctness of stylus pressure, and extent of wow. An excellent set of instructions supplied with the record greatly extends its usefulness.

SOUND HANDBOOK

(*from page 32*)

amplitude modulation occurs twice per low-frequency cycle. The intermodulation frequencies in this case will consist of the sum of and difference between the high-frequency signal and double the frequency of the low-frequency signal, the latter figure representing the rate of modulation of the envelope.

Harmonic distortion is described quantitatively as the ratio, in percentage, between the r.m.s. amplitudes of the spurious harmonics and of the fundamental. Each order of harmonic may be listed separately, or the total distortion may be expressed in a single percentage representing the graphical addition of all component orders. The total will thus be less than the arithmetic sum of its various parts. Higher orders of harmonic distortion have an especially great irritation value.

Figure 8-6 is a table listing the maximum harmonic distortion allowed broadcast stations by the FCC. Modern high-quality recording equipment, pickups, and amplifiers in particular are able to do considerably better than the FM performance standards. These standards are still too high for loudspeakers, however. Speaker distortion data is rarely published, but when such data does appear it reveals the fact that the very best and expensive loudspeakers create distortion, at certain lower portions of the frequency spectrum and at moderately high power levels, of the order of 5 per cent.

Intermodulation distortion is also indexed as a percentage ratio, that between the amplitude of the unmodulated high-frequency carrier and the amplitude of the low-frequency modulating envelope. This is the same figure as the percent of modulation referred to in AM transmitter work. The ratio between harmonic and intermodulation percentage is not fixed, and depends upon factors such as the signal frequencies, their relative intensities, and the type of non-linear distortion involved, but usually the intermodulation distortion is between three and four times the harmonic distortion. It will be seen that multichannel systems, in which the low and high frequencies are not allowed to pass simultaneously through the same distorting system, discriminate against the intermodulatory effects of harmonic distortion.

Power Capability

The generally accepted goal of a home reproducing system, as far as power capability is concerned, is to be able to create intensity levels of sound in the living room equal to the intensity level of sound at a good seat in the concert hall. This is not as overpowering an intensity as it might at first seem; the relatively quiet coughs and sneezes of one's neighbor at the concert hall may be heard with ease during the performance.

The acoustical output of the reproducing system is spread over a much smaller area, and is therefore able to recreate concert hall intensity levels with much smaller amounts of power. The acoustical power of a seventy-five piece orchestra may approach peak instantaneous values of 75 watts, which, with a typical home speaker system of 5 per cent efficiency, would require an instantaneous electrical power capability of 1,500 watts to reproduce. We are obviously not concerned with absolute values of power, however, but with sound intensity, in watts per square centimeter (watts/cm^2.)

The walls and other surfaces of a room are partially reflecting, so that sound directed against these surfaces will not disappear immediately after the source stops, but will be reflected back and forth, losing part of its energy with each reflection. The sound will take a certain amount of time to die away; the time required for the steady-state intensity to *decay* to a value one millionth of the original (60 db down) is called the *reverberation time*. It is directly proportional to the volume of the room and inversely proportional to the total absorption present.

The acoustical power required to create a given intensity level is inversely proportional to this reverberation time, because new sound energy being pumped into the room finds some of the old sound energy still working. Thus the absorptive materials in a room require a source of sound to radiate greater power for the same intensity level. *Figure* 8-7 is a chart indicating the acoustical power required to produce intensity levels of 70 to 100 db (above threshold) for various auditorium or room volumes with typical reverberation characteristics. Since the desired concert hall levels may approach peaks as high as 100 db (*Fig*. 8-1), a room 10 × 15 × 20 ft. would require a maximum acoustieal output of 0.45 watts. With a loudspeaker system of 5 per cent efficiency, this output would call for an amplifier with a maximum power capability of 9 electrical watts. Larger rooms, a great deal of sound absorbent furnishings, and less-efficient speakers increase this requirement.

The power rating is determined by the number of watts that can be drawn from the system without distortion exceeding a specified amount. This rating tends to drop at the frequency exremes. *Figure* 8-1 shows that such a deficiency may be offset by the reduced intensity levels of orchestral music in the end frequency regions.

Dynamic Range

The dynamic range required for the reproduction of orchestral music with perfect subjective fidelity is about 70 db (see *Fig*. 8-1). The upper limit of the dynamic range of reproducing equipment is determined by the power capability of the system (in the case of disc recording by the maximum groove deviation allowable); the lower limit is determined by the level of noise, which would mask low-intensity sounds. Older records used a dynamic range severely limited by considerations of over-cutting of grooves and of surface noise, and the dynamic range of AM broadcasts is limited at each end by modulation percentage and signal-to-noise ratio respectively. These program sources therefore have a restricted range, often 40 db or less.

The use of expander circuits, which increase the gain of the amplifier to loud signals and restore some of the dynamic range lost in the original compression, was fairly common a few years ago. Although the characteristics of the expansion could not be perfectly matched to those of the original compression, the better circuits afforded considerable over-all improvement. Today, because of the extended dynamic range of records and of FM broadcasts, approaching the full requirement, the use of expanders has decreased.

Noise Level

The spurious noise components introduced into the output of reproducing systems include hum, thermal noise, microphonics, record surface noise, and turntable rumble. The annoyance value of noise is not determined exclusively by its relative amplitude to the signal, but also by other of its characteristics, particularly frequency content (due to the increased hearing sensitivity at certain parts of the spectrum described by the Fletcher-Munson effect). Noise of very low frequency may be inaudible but may create intermodulatory products in interaction with the signal. Noise level is normally described in terms of the ratio to the maximum signal amplitude, as this rating is more accurately indicative of annoyance value in a given situation than a rating in terms of absolute power would be. *Figure* 8-6 shows the maximum power level of noise allowed broadcast stations by the FCC. When the noise level is 60 db below the maximum signal power the absolute value of the hum in microwatts is equal to the value of the signal output in watts.

Due to the influence of acoustical conditions, hearing characteristics, and other such factors on the irritation value of noise, a listening test may in some situations be superior to a quantitative measurement, unless the quantitative results are properly weighted.

Binaural Reproduction

Sound which originates from an orchestra approaches the listener from several directions. The path length to each ear is therefore not the same, and in addition the waves are diffracted around the head, so that the version of the sound which each ear receives is slightly different in time, phase, intensity, and even timbre (this last due to the frequency discrimination of diffraction). Experience has enabled our perceptive mechanism to interpret these differences in terms of directional location of the source or sources. Monaural reproduction from a single reproducing source eliminates this factor of realism, and has a definite effect on the illusion created, although there may be differences of opinion on the importance of the musical values associated with the loss.

The lost illusion may be restored, to a greater or lesser degree depending upon the method used, by stereophonic or binaural reproduction. A practical binaural system that is perfect can only work through earphones. Two recording microphones are mounted as though each were one human ear; the sound received by each is recorded separately, and each recorded channel is played back into its corresponding left or right ear-piece. In this way each ear receives a version of the music with just those characteristics that the left or right-side version would have contained in the concert hall.

Since earphones are much less convenient than loudspeakers, most practical stereophonic systems are designed for speakers. Each channel no longer represents the sound received by each ear, but the sound coming from the left or the right of the hall, all of which is received by both ears. This arrangement, for accuracy of stereophonic effect, would require many channels and a careful arrangement of the speaker array calibrated to the original layout of instruments. Such a procedure is obviously impractical except for the theater, and typical stereophonic systems use two channels and two permanently mounted loudspeakers. The results are a compromise but nevertheless afford an appreciable increase in "liveness."

Binaural demonstrations with earphones are always much more dramatic than those with louspeakers. The reason for the greater illusion is described above, but there is another factor that increases the contrast between the monaural and binaural sound. Single-channel reproduction through earphones creates an artificial and impossible acoustical condition; each ear receives the identical sound at the same instant. Because of the fact that the normal left-right differences associated with diffraction around the head and reflection from room surfaces are eliminated when the ears are covered, such reproduction is unusually "flat", even when compared to that of a single loudspeaker. The contrast is thus widened by a particularly weak monaural standard.

NEW LITERATURE

• **Engineering Products Department, RCA Victor Division,** Camden 2, N. J. has produced a full-color sound film titled "You Are the Producer," which demonstrates and explains how educators, industrial training groups, home movie makers, etc., can record their own sound on 16-mm film at nominal expense and without laboratory processing. The 12-minute film features the RCA "400" magnetic recorder-projector in an interesting dramatic sequence based on the ability of the unit to record on magnetic track, play back, erase, reproduce both optical and magnetic track, and operate at sound or silent speed. Information concerning use of the film for showing before interested groups may be obtained from any RCA visual products distributor, or by mailing an inquiry to the address shown above.

• **Thordarson-Meissner,** Mt. Carmel, Ill., is now distributing the company's new general catalog No. 53-A, a complete listing of all items manufactured under the Meissner trade mark. Many new products are shown for the first time, including a 10-watt high-fidelity amplifier and a transmitter kit for novices. A special section is devoted to receiver and amplifier kits. Copy will be mailed free for the asking.

• **General Electric Company,** Tube Department, 1 River Road, Schenectady 5, N. Y., announces availability of a new booklet describing the essential characteristics of G-E Five Star miniature and subminiature high-reliability tubes. Included in the data are the specific differences between individual Five Star tubes and their standard-tube counterparts. When writing specify Publication ETD-548-A.

• **McIntosh Laboratory, Inc.,** 320 Water St., Binghamton, N. Y., will mail free a new 32-page booklet titled "Lost Instruments," which makes use of cartoons to explain the whys and wherefores of high-quality audio equipment. Shown graphically are the meanings of such technical terms as intermodulation, harmonic distortion, band width, etc. An informative publication for both beginners and veterans in the audio field. Write for it.

• **Gustin-Bacon Manufacturing Co.,** 210 W. 10th St., Kansas City, Mo., is introducing Ultrafine, a new thermal and acoustical insulation, with a handsome 4-page folder which includes illustrations of the product in use, as well as tables listing thermal and acoustical efficiency. Ultrafine is made of extremely fine, blown glass fibres, and can be obtained in a variety of facings.

• **Automatic Electric Co.,** 1033 W. Van Buren St., Chicago 7, Ill., describes and illustrates a complete line of telephone-type relays including hermetically-sealed subminiature and plug-in types, in a colorful new 12-page brochure designated as Circular 1702-A. This is one of the finer examples of industrial publishing to cross this desk in many moons. If you use small relays, you will find a copy of great value.

Employment Register

Positions open and Available personnel may be listed here at no charge to industry or to members of the Society. For insertion in this column, brief announcements should be in the hands of the Secretary, Audio Engineering Society, P. O. Box 12, Old Chelsea Station, N. Y. 11, N. Y., before the fifth of the month preceding the date of issue.

★ Positions Open • Positions Wanted

★ **Audio Systems Engineer.** Opportunity in new Audio Systems Section to create new products in professional and reproducing equipment. Telephone Mr. Marshall, INdependence 3-2100 days, FLoral Park 2-0578 evenings. Fairchild Recording Equipment Corporation, 7th Avenue and Powell's Cove Road, Whitestone, N. Y.

FLEXIBLE CHANNEL

(from page 29)

because the final product would be on optical film and after making comprehensive listening tests it was decided that normal optical cut-offs would be useful for magnetic recording also. Due to the increased amount of recording being done, everything is first put on magnetic and then transferred to optical and held until the picture "answer" prints are delivered. In this way the advantage of magnetic is utilized as a safety, as well as for a temporary reference; it is erased after the "all clear" is given. Some time ago, the inventory of magnetic film increased to such an amount—because of the increase of jobs waiting for answer prints—that the system was later modified and magnetic film is erased and made available for re-use immediately after getting the "go ahead" from the editorial department when they are satisfied they have the proper footage to cover their needs.

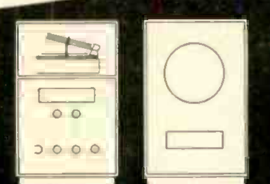
As far as the actual operation of the channel is concerned, it was made as straightforward an operating chore as possible so as to eliminate mistakes, floundering for switches, and so on. Therefore, the entire unit is practically left to the Mixer for operation and he has all the switches and patch bays at his fingertips. In operating procedures with the channel, key K_4 permits the mixer to listen to the signal at the recording amplifier buss or to the signal that is actually recorded on the magnetic film. No change in equalization is needed —the Mixer has merely to throw a switch. This is used for all original scoring so that the Mixer himself can be sure that he "has" the take—some jobs not permitting time for a playback. This type of monitoring was not provided in the re-recording operation because of the trouble caused by the time delay. Incidentally, the time delay does not seem to bother the Mixer during direct sound recordings while watching the speaker's lips. It is felt that knowing the take is on the film and in good shape seems to offset the lack of synchronized monitoring. The talkback and monitor amplifiers are used in a conventional fashion but are interlocked by key K_1 so as to eliminate any possibilities of feedback while using their circuits. Relay K_3 is also used as an interlock to prevent the same difficulty when the microphones are "hot" and a playback is needed on the stage. Key K_6 is used mainly for playing back any original scoring for checking purposes and automatically sends the signal into the linear and flat channel, thence to the playback amplifier and speaker. This is in line with the wanted design in that the Mixer does all the switching involved with the exception of actually rolling the recorders and reproducing equipment. This makes it possible to utilize the services of a relatively inexperienced recordist without having to keep a constant supervising vigil.

Motor and Signal Circuits

Referring to *Fig.* 2—the schematic showing the motor and signal circuits—it will be seen that there are four "ready light" positions available—two of which are duplicated in the re-recording room. These positions utilize a holding relay that automatically lights a lamp of the proper color at each of the positions. When the machines start to roll, the power is lifted from the circuit and the lights automatically go out. It was originally thought that—because if a ready light was once put on, the operator could not turn it off if he changed his mind—there might be some trouble. However, this took place only in isolated cases, and then only because of unfamiliarity with the system—a condition encountered occasionally with new or inexperienced personnel. All that is necessary if the operator has to change his signal is for him to use the telephone which is normalled between all sound-interested places. When the units start to roll, the ready lights are released and they go out. In case of trouble, the projection rooms have their own selsyn control, used only for stopping. If the machine room has trouble, the operator makes sure the projection lamp is out—by using the telephone—before stopping. For quick emergencies, the talkback microphone can be used. In addition, each selsyn control box has lights showing when it is in a locked position and when rolling so that all interested personnel know exactly what is operating. The "lock" lights are convenient in that they are a signal for the machine men and the projectionist to check their respective motors to make sure that they are in lock and not stalled on a "high spot." This completely eliminates the possibility of having a runaway machine. The solenoid that disconnects the ready lights is energized by the selsyn control boxes during dubbing and by a separate push button during original voice or music scorings. The same solenoid also energizes the warning light system and automatically rings a three-second bell at the start of recording and sounds a three-second buzzer at the end. These unrelated audible signals have been found sufficient to alert any personnel completely while a "take" is on. Provision is also made to control any a.c. machinery needed by means of the same solenoid relay. A rachet-type relay was selected so as to draw current only when the contacts are being moved from OFF to ON, or vice versa.

The installation has proved itself time and time again, with very few troubles, to operate in a straightforward and efficient manner in carrying out its sound recording duties, and has come up to the expectations of all concerned.

The author wishes to take this opportunity to thank everyone who has contributed directly or indirectly in the past five years to his film-recording "education." Particular thanks go to Mr. H. D. Bradbury and his colleagues of RCA Victor Film Recording Division for their helpful advice and assistance.



ZERO IMPEDANCE

(from page 21)

The output transformer should match the loudspeaker to this load. A good compromise for R_1 is 2 to 4 times the magnitude of Z and the time constant R_1C_1 should be approximately 150 microseconds.

The value for Z used by the author was low, being around 5,000 ohms; R_1 was 15,000 ohms and C_1 was 0.01 μf. These values are not critical.

For R_1C_1 time constant of 150 microseconds, the reactance of C_1 is equal to R_1 at approximately 1000 cps. Consequently the power loss in the resistor is low below this frequency. In music, very few fundamental tones occur above this frequency which corresponds to two octaves above middle "C" on the piano. Even though some available power is lost at higher frequencies, it is not a serious matter.

In the usual 6V6 class "A" power amplifier, the average screen current and the average plate current both increase when a signal of sinusoidal or symmetrical waveform is applied. If the signal is keyed on and off and a d.c. milliammeter placed in the plate circuit, the meter fluctuates wildly. If instead of maintaining a constant screen voltage, a particular value of screen dropping resistor R_3 is chosen, it is possible to minimize this fluctuation in d.c. plate current provided the screen bypass capacitor is removed. The action is simple. In the presence of signal, screen current increases. The resulting increased voltage drop in R_3 causes a reduction in screen voltage just sufficient to provide the required compensation In order to obtain instant action, the screen-bypass capacitor must be removed. The removal of this capacitor results in about 10 per cent reduction in voltage gain, and a loss in screen filtering.

The use of direct-coupled voltage feedback eliminates the need for a large blocking capacitor and ensures proper operation of the feedback circuit at the lowest audio frequencies. The current through the feedback resistor R_{f1} must come through the output transformer, which is a disadvantage. Resistors R_{f1} and R_{f2} should preferably be wire wound.

Positive current feedback is obtained from a portion of the cathode bias resistor R_{k2} of the 6V6, as shown in *Fig. 1*. R_{k2} is a wire wound potentiometer, and serves as a control to adjust the output impedance. This potentiometer may be replaced by a 200-ohm resistor and the current feedback taken across the full 200 ohms.

The positive feedback control is adjusted in the following manner so that the tube presents zero impedance to its plate load. With the loudspeaker connected, an a.c. voltmeter is connected across the primary of the output transformer with moderate signal applied to the amplifier. A resistor of 5000 to 10,000 ohms (not critical as to value) is then shunted across the transformer primary. The feedback control is adjusted to a point where there is no change in output voltage as this resistor is connected or removed. With full 200 ohms in the feedback control, the output voltage actually increases when the resistor is connected across the transformer, indicating a negative-impedance source. It was found by measurement that the source impedance in the author's amplifier remains zero from 20 to 20,000 cps. It was not checked above this frequency. The drop off below 20 cps is due to the coupling capacitor C_2.

It should be noted that a blocking capacitor cannot be used in series with the negative feedback resistance R_{f1}. If one were used, the negative feedback would become ineffective at some low frequency, yet the positive feedback would still be effective, and low-frequency oscillation or motorboating is likely to occur.

This circuit has been used by the author for several months, with much enthusiasm. The high frequencies are clear and crisp and piano music is reproduced with excellent clarity. The power output is ample for low-level applications.

R-J SPEAKER

[*from page 20*]

Construction

The internal construction of the R-J speaker enclosure is shown quite clearly in *Fig.* 4. All wood used in the construction is ¾-in. plywood, with the external surfaces veneered with suitable hardwood to provide an attractive finish. The dimensions shown will suffice for either 12- or 15-in. speakers, since the speaker size is not critical. The principal requirement is that the free-air cone resonance be relatively low, and the suspension is "soft" enough to provide a high compliance. In the cabinet of *Fig.* 4, a net internal volume of 4500 cu. in. is provided, exclusive of the internal padding. It will be noted that the speaker board is spaced from the frontal board by such a distance as to reduce the peak caused by the resonance of the enclosure so that the sound output is flat, essentially, from a frequency of 150 cps, let us say, down to 50 cps or less. As previously mentioned, the spacing between the speaker board and the frontal board determines the Q of the enclosure, and effective eliminates the peak due to the Helmholtz resonator effect.

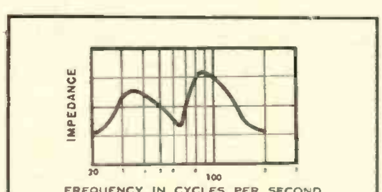


Fig. 5. Impedance curve of the R-J enclosure with the same type of speaker as used in the cabinet of Fig. 1.

The opening in the speaker board is the same as the free cone area of the speaker used—10½ in. for a 12-in. speaker and 13½ in. for a 15-in. speaker. The opening in the frontal board was shown to have an area of 64 sq. in.

If the enclosure is to be used as a woofer exclusively, a rectangular slot—as shown in *Fig.* 4—would be perfectly satisfactory, and adequate performance would be obtained over the frequency range from the lowest transmitted frequency up to 400 or 500 cps. However, if the enclosure is to be used for a wide-range speaker, without the addition of any other speaker or horn for the high range, then the frontal opening must be modified. While the shape of the opening must be determined specifically for each individual cabinet type, it may be described roughly as resembling a "skinny lemon." It may be approximated by laying out a diamond-shaped opening with its major axis about 25 deg. from the vertical—at an azimuth of 025, for the nautical-minded—and with its minor axis at an angle of approximately 15 deg. from the horizontal, or at an azimuth of 295. The minor axis should be one half the length of the major axis, and the total area should be the calculated 64 sq. in. Then, after the diamond shape is laid out, round off the obtuse corners, and extend the acute corners so that the "skinny lemon" shape is attained. From this description, it is obvious that it would be difficult to prescribe the actual opening, but bear in mind that the low-frequency performance is a function of the enclosure volume and the area of the frontal board, and the high-frequency performance is a function of the shape of the frontal opening. The spacing between the speaker board and the frontal board adjusts the Q so as to avoid a peak in sound output throughout the range from 50 to 100 cps.

From the foregoing, it is seen that the low-frequency performance is not as dependent upon the speaker as it is with some other types of housings. It is only important that the free-air cone resonance be relatively low—from 40 to 70 cps. Since the object is to resonate the enclosure somewhat above the cone resonance, and since the volume and the size of the frontal opening determine the resonant frequency, it is readily possible to design a satisfactory cabinet for speakers smaller than the 12- or 15-in. models for which the cabinet of *Fig.* 4 was calculated. This has been shown by the performance of the 8-in. enclosure, which is 23½ in. long, 11 in. high, and 10 in. deep. The design of this cabinet is the same as for the larger one, being modified only by the differences in size and resonant frequencies.

Comparative Measurements

To show the performance of the system, and to evaluate it in comparison with a conventional bass-reflex box, the equipment shown in *Fig.* 6 was set up. The amplifier is a modified Williamson employing two 5881's in the output stage, and with 30 db of feedback. The internal output impedance of the amplifier was approximately 0.5 ohms. The

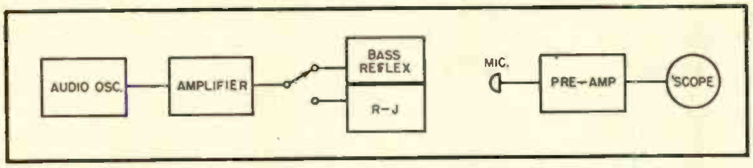


Fig. 6. Block diagram of the test set-up used to compare acoutsic outputs of the two types of enclosures.

two enclosures used were the bass-reflex model of *Fig.* 1 and the R-J of *Fig.* 4, both being used with a moderately priced 12-in. speaker with an 8-ohm voice coil, and a free-air cone resonance of 63 cps. Both cabinets were placed side by side with the microphone on a horizontal axis through the two cones and in line with the two adjacent sides of the enclosures. The output of the microphone was fed through a preamplifier to a 5-in. 'scope. The switch permitted feeding the output of the amplifier to either enclosure and the microphone output could then be observed on the 'scope screen. Thus all variables have been eliminated, with the possible exception of differences between the two speakers—although both were identical as to type and manufacturer. *Figures* 7 through 12

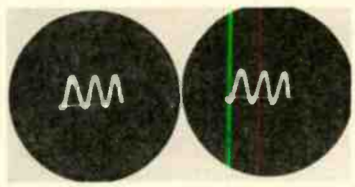


Fig. 7. Acoustic output from the two speakers cabinets as indicated on the 'scope at a frequency of 110 cps. Left, bass reflex; right, R-J.

show the patterns obtained, none of the controls being changed except the main frequency dial of the oscillator and the selector switch. In all of these figures the bass reflex cabinet is shown at the left and the R-J enclosure is shown at the right.

Figure 7 shows approximately the same sound pressure from both cabinets at 110 cps. *Figure* 8, at 84 cps, corresponds approximately to the cabinet air resonance for both enclosures, as shown by the impedance curves of *Figs.* 2 and 5. Note the higher output of the bass reflex, resulting in the characteristic "boominess." The R-J output is only slightly increased, and even this could be eliminated by a slight decrease in the spacing between the frontal board and the speaker board.

In *Fig.* 9, taken at 55 cps, both

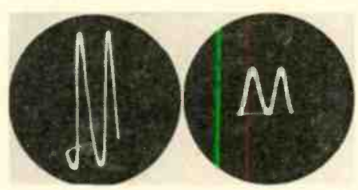


Fig. 8. Acoustic output at 84 cps. Left, bass reflex; right, R-J.

speakers exhibit approximately the same output, but in *Fig.* 10 at 48 cps, the bass reflex is beginning to exhibit traces of frequency doubling. The R-J is still entirely fundamental. At 44 cps, shown in *Fig.* 11, the bass reflex has an output which is almost entirely composed of double the frequency of the applied signal—which means almost 100 per cent second harmonic distortion. The difference in pitch is plainly evident upon switching between the two, the bass reflex sounding one octave higher than

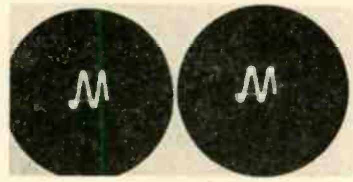


Fig. 9. Acoustic output at 55 cps. Left, bass reflex; right, R-J.

the R-J. *Figure* 12 shows that the bass reflex is still frequency doubling, while the R-J—although somewhat down in output—still maintains the fundamental tone.

The ability of the R-J system to handle power without frequency doubling at the very low frequencies is due in the main to the air loading on the speaker cone which is the result of the design. This loading is also beneficial in increasing the transient damping of the speaker as well as the power handling capacity. Transient damping which cuts down spurious speaker vibration of

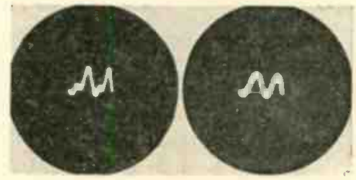


Fig. 10. Acoustic output at 48 cps. Note beginning of frequency doubling from the bass reflex at the left; the R-J is still fundamental.

"hangover" after the signal has ceased may be demonstrated in a simple test. If a small direct current from a battery is keyed on and off, a "click" will be noticed on circuit make, and either a "click" or a "bong" on circuit break, depending on the damping.[5] It will be noticed particularly with the bass reflex, which exhibits a definite "click-bong" as the circuit is made and broken, while the R-J shows its transient response to be better by responding with a "click-click" for the same treatment.

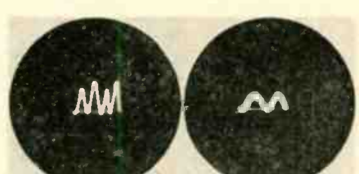


Fig. 11. Acoustic output at 44 cps. Output from bass reflex is seen to be almost 100 per cent harmonic, due to frequency doubling.

Conclusions

From these comparisons, it may be concluded that although it is based on a resonant principle, the R-J enclosure is a non-resonant system. The response with a 12-in. cone having a free-air resonance of 63 cps is smooth down to 50 cps, and has usable but attenuated output down to 30 cps. It is also shown

[5] Benjamin B. Drisko, "Getting the most out of a reflex-type speaker." AUDIO ENGINEERING, July 1948.

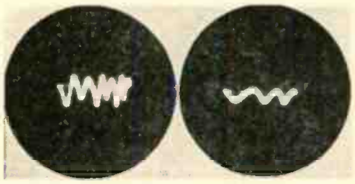


Fig. 12. Acoustic output at 32 cps. Although slightly lower in output, the R-J—at the right—is still fundamental in character.

that the R-J provides air loading which is instrumental in preventing frequency doubling and tripling, and maintains fundamental tones with a 12-in. speaker down to 30 cps. In addition, the R-J offers excellent speaker damping with the result that transient response is maintained.

BINAURAL RADIO

(from page 14)

response than AM assure positive support from this second channel.

Those who want binaural reception must own an FM receiver—which means an increase in sales and production of FM or AM-FM sets. This is a step in the right direction to help establish a more solid commercial value for the FM station. This is a very real need, because of receiver manufacturers' cutting back or stopping production of complete package units. If this transition to binaural reception can grow rapidly enough, it may increase the number of FM set owners, and many of them will become converted to the FM monaural stations. Enough of such activities might prevent some FM station failures.

During the transition from monaural broadcasting to various new levels of binaural programming, many listeners will find that the advantages of listening to two radios will broaden the area of sound reproduction, getting away from the single-spot source of sound. One-receiver listening eventually will swing back to FM, which again will build up the value of the FM service.

The FM and the AM reception is in no way reduced in efficiency by using one receiver monaurally, while the value for both services is increased to the listener when used together for binaural reception. The power consumed at the two transmitters costs the same regardless of the studio pickup technique—monaural or binaural. However at the listening end the two receivers, when tuned to two stations transmitting binaurally, will produce more usable listening volume, allowing the volume controls on each receiver to be set much lower than required for monaural listening. This is a gain in efficiency yet to be accurately evaluated, but it definitely exists.

At Troy, New York, Rensselaer Polytechnic Institute operates stations WHAZ and WRPI for experimental and public service programs. Both of these stations use AM, one on 1330 kc and the other on 640 kc. Listeners in this area are now enjoying binaural programs with two AM receivers tuned to the two frequencies. This method of binaural broadcasting is permissible under FCC regulations when the station which originates the program, either live or recorded, gives permission in writing to another station for simultaneous or delayed broadcasting. Co-operation between two AM or two FM stations in a territory served by both could be the means of binaural broadcasts. As good-will builders and long-range public relation service it has great value.

In television and movies, another handicap is added to nature's facilities by using a single-lens sight pickup on the camera. These two handicaps add to a condition where both sight and hearing are fatigued more rapidly than in nature where two ears and eyes are used normally, as at the opera, stage, or concert.

Binaural radio sound reproduction provides nature's two-ear listening facility and when stereo television with binaural sound becomes available fatigue will be greatly reduced. Many observers agree that (two ear) binaural radio broadcasts will fatigue us less rapidly and keep us alert to the program—including the sponsor's message—for a longer period. This may help the sponsor make a deeper impression on the listener against the chatter of those assembled at home to hear a favorite program, even though these listeners are using it for atmosphere background.

Activities in 1952

In late October 1952 binaural broadcasting in New York was introduced at the Audio Fair, and its impact is still being felt. Station KOMO in Seattle in the same month did a successful broadcast which struck a high level of local appreciation.

Station WJR in Detroit used its AM-FM stations to produce two binaural broadcasts each week for eight weeks during July and August. These broadcasts were enthusiastically received by hundreds of hi-fi fans with their wider range AM-FM combinations, and a big audience of listeners using package AM and FM receivers. These concerts featured the Detroit Symphony Pops Orchestra under the direction of Ed Werner at the Jefferson Beach open-air music center. Detroit newspapers carried a continuous one-column cut showing a diagram for a receiver setup at home. Spot announcements daily kept the audience growing. Future plans are soon to be announced.

Station WQXR in New York agreed early in September to program an AM-FM broadcast for the banquet entertainment at the Fourth Annual Convention of the Audio Engineering Society on October 30. This finally resolved in two binaural broadcasts sponsored by Magnecord, Inc. on October 29 and 30. The October 29 broadcast used a binaural Magnecorder and a completely taped show. The program on October 30 was produced by WQXR's musical staff with a live 28-piece symphonic ensemble and two voices. In the ballroom of Hotel New Yorker, the banquet guests listened to the radio reproduction from Electro-Voice Patrician speakers fed by Fisher Radio Corporation tuners and amplifiers. Norman Brokenshire, well-known announcer, gave his hearty approval from the studio where he was listening "off-the-air" to two RCA table model receivers. This program was the first known binaural network broadcast, as it was relayed by FM radio to WDRC-FM on Mount Meriden in Connecticut, and the AM was piped by telephone line to WDRC studios in Hartford, Connecticut.

Both WQXR and WDRC were swamped with phone calls from a large audience, and requests for more binaural broadcasts followed by mail. These requests were fulfilled on November 9 when the Harvey Radio Company of New York started a 52-week series of one-hour binaural broadcasts in co-sponsorship with Magnecord, Inc. The program is the regular WQXR

String Quartet from 6 to 7 p. m. each Sunday. This show is well produced and the effect is thoroughly realistic.

Beginning January 3, WQXR will present each Saturday from 4:30 to 5:00 p. m. through January, February, March, and April, a series of organ recitals by outstanding members of the American Guild of Organists. These programs will originate from Temple Emmanu-El under the direction of Dr. Hugh Giles, Chairman of the Guild and organist at Central Presbyterian Church in New York.

WDRC in Hartford is revamping one of its studios (20′×28′×16′) for binaural programming. WDRC is owned and operated by F. M. Doolittle, mentioned earlier as the daddy of binaural broadcasting.

WNYC, New York City's municipal station, arranged to have Mayor Impelliteri give his Christmas Day message binaurally. It is also planned to broadcast an intensive binaural series from February 12 to 22 with the entire annual American Music Festival, including the United States Army Band and the New York Philharmonic Orchestra directly from Carnegie Hall. As a series, these binaural broadcasts will be the most pretentious yet to be performed, and should go far to establish binaural radio programming.

Station WGAR in Cleveland has recently done a series of nightly 15-minute programs of binaural tape with encouraging audience reaction. One listener said his dog ran back and forth from one radio to the other to catch the ping-pong ball reproduced binaurally during the program. Intensive plans are under way for both live symphony and tape shows to satisfy a big audience.

Station WCAE in Pittsburgh produced a series of successful binaural broadcasts in November, using tape and live orchestra. The station plans to do a great deal with binaural programming in 1953.

These brief comments about binaural broadcasts and plans for the stations mentioned were picked up in telephone conversations with program personnel at each station early in December. The personal enthusiasm of each individual reflects the spirit necessary to bring healthy changes into broadcasting. This article was not intended to be technical. For those interested, reference should be made to the excellent article "Stereophonic Sound Reproduction" by James Moir of England, which appeared in the September issue of Æ.

Jack Seligman

"We're the repair men for your binaural sound system."

AUDIO TRANSFORMER

[*from page 44*]

represents 2 db, and 99 per cent represents 0.23 db.

The use of the charts in *Figs*. 5 and 6 to determine efficiency is illustrated by *Fig*. 7. *Figure* 5 is used to find the maximum efficiency of which a transformer is capable, and the relation between the actual operating impedance of the transformer and that producing this maximum. *Figure* 6 then gives the actual efficiency from this information. r_w is the total winding resistance referred to the winding chosen for making the calculation; R_p is the shunt loss due to the core, referred to the same winding, and measured at a mid-frequency of, say 1000 cps. (Although somewhat lower frequencies are often taken for mid-frequency for other audio purposes, 1000 cps gives a fairer picture for audio transformer efficiency). An impedance which is the geometric mean of r_w and R_p results in maximum efficiency, and this maximum efficiency is given by the η scale on *Fig*. 5. At the same time, reference in the manner shown in *Fig*. 7 gives a value of Z_{opt}/Z, indicating by what ratio the actual impedance, Z, at which the reference winding is operating, differs from Z_{opt}, which gives this maximum efficiency.

The maximum efficiency point is not always the best impedance at which to work a transformer, because frequency response must be considered as well, and in smaller models harmonic generation may also affect the consideration.

Having found what the efficiency is for one particular impedance, the construction of the charts avoids the necessity for going through the whole procedure again for another impedance; the efficiency for the new impedance may simply be read off from a different point on the same curve of *Fig*. 6.

Harmonic Generation

This is a relatively simple calculation, so no special chart has been provided for it. The previous article gave a method of measuring harmonic content as a percentage of fundamental magnetizing current. Magnetizing current at the frequency and amplitude concerned can be expressed as a shunt impedance referred to the primary. The source impedance and load impedance referred to the primary can also be expressed as a combined shunt impedance. To find the effective harmonic percentage generated, the percentage measured is then divided by the ratio between the shunt impedance due to magnetizing current and that due to external circuit impedances. For example, if the measured percentage is 15 per cent, the shunt impedance due to magnetizing current 50,000 ohms, the plate resistance 20,000 ohms, and the referred load resistance 5000 ohms: the external shunt impedance is 4000 ohms, and the effective

harmonic generation, $\frac{15 \times 4}{50} = 1.2$ per cent.

Input Transformers

Insertion loss in an input transformer is usually unimportant compared with its frequency characteristic and other properties. It is normally direct coupled, but with no d.c. polarizing. Due to low signal levels, working flux densities in the core are extremely small, and the effective primary inductance varies over a considerable range with variation of signal level. Inductance should be sufficient to maintain good response, even with its smallest value. If it is not sufficient to do this, the only remedy is to reduce the working impedance by additional shunt resistance, which will also add insertion loss—at a point it can ill be afforded; so generally a compromise is necessary, or a better transformer.

Its h.f. response is generally determined as to range by the impedance to which the source is matched by its step-up, and the input capacitance of the tube it feeds. Sometimes an h.f. peak appears, as will be shown by use of *Fig.* 1. In such cases increase in input capacitance pulls down the frequency of peak, but usually increases its height; the best remedy is an appropriate value of shunt *R* across the secondary. If this restricts the range more than is desired, tube input capacitance must be reduced as well, by the use of a little inverse current feedback (leaving off the cathode by-pass capacitor). The only alternative is another transformer with less step-up.

If the whole chain has a deficiency in highs that can be corrected by peaking, the input transformer is a good place to introduce such correction, because it will boost the highs in the signal before the thermal noise of the first stage is added to it.

Shielding against hum pick-up is not one of the properties within the scope of this article, but it is important for some input transformers, and so should be mentioned. Some reduction can always be effected by orientation, but if shielding is totally inadequate the only answer is a better shielded job.

Interstage Transformers

Direct-coupled types carrying d.c. polarizing usually have larger leakage inductance and self-capacitance than types without provision for d.c. polarizing. If low frequencies are deficient, the best remedy is to shunt the primary with a suitable resistor. This will bypass some of the polarizing current, thereby increasing inductance slightly, and at the same time reduce effective source impedance. But this shunt resistance may produce a peaking effect in the h.f. response. As with input transformers, this may be overcome by suitable shunt resistance across the secondary. The charts can be used to find the best combination more readily than protracted trial and error, taking a frequency run each time a change is made.

Parallel-fed types of interstage transformer may cause peaking at one or both ends of the audio spectrum. In general it is best to avoid peaking at the l.f. end, aiming at a value of L/CrR greater than unity, so that variation of inductance with level produces a minimum variation in response. However, going too far in this direction may restrict the l.f. range more than is desired; so the charts may be used to find the best compromise, using two different values of inductance (extremes of variation with level) to see what effect signal level can have on response.

The h.f. end will be easier to handle, because with most parallel-fed components there is more in hand. The limit to step-up is generally set in the same way as for input transformers.

Some points should be noted here that help in getting just the right effect at both ends of the spectrum at once. Resistance across the secondary has the effect of reducing tendency to peak at both ends, although maybe in different degrees, according to relative values of L/CrR for each end. Varying the coupling resistor also has a similar effect at both ends, but possibly different in degree. Usually if the secondary shunt resistor has more effect at one end, the value of coupling resistor will have more effect at the other. But a resistor in shunt with the *primary* (after the coupling capacitor), has opposite effects at each end: at the l.f. end it is the same as an equivalent referred value shunted across the secondary, and so reduces the tendency to peak; but at the h.f. end it shunts the source resistance, reducing the value of r, so increasing the tendency to peak.

Thus adjustments of coupling resistor, coupling capacitor, shunt primary resistor, and shunt secondary resistor, afford a considerable variety of possibilities with the parallel-fed type of interstage transformer. Exploration of these possibilities to find the best combination can be greatly expedited by use of the charts.

Output Transformers

In output transformers everything takes on different proportions. The transformer is step-down, and most circuits—even using tetrode or pentode outputs (inverse voltage feedback being invariably used)—have an effective plate resistance that is but a fraction of the referred load resistance; so peaking at h.f. is impossible. Instead, harmonic generation and efficiency are now the features to watch. If harmonic generation at the lowest frequency of interest is kept low, l.f. response also will usually be taken care of. The charts in *Figs.* 5 and 6 will be useful in working out efficiency, particularly to determine whether a transformer designed for operation between, say, 7000 and 5 ohms, will serve satisfactorily between 10,000 and 7 ohms, or between 5000 and 3½ ohms.

The remarks here made about output transformers also apply to loudspeaker and line-matching transformers.

LEADING WEST COAST AUDIO REPRESENTATIVES

The representatives listed below are for the use of dealers and distributors only. The Hobbiests and Experimenters are requested to contact their dealer for information.

A MONTHLY SUMMARY of product developments and price changes of radio electronic-television parts and equipment, supplied by United Catalog Publishers, Inc. 110 Lafayette Street, New York City, publishers of Radio's Master.

These REPORTS will keep you up-to-date in this ever-changing industry. They will also help you to buy and specify to best advantage. A complete description of most products will be found in the Official Buying Guide, Radio's Master—available through local radio parts wholesalers.

Miscellaneous Radio, TV and Electronic Parts

BIRNBACH RADIO—Introduced No. 6261 at $1.14 net and No. 6263 at $1.41 net, both bat-handle center off switches of molded bakelite, rated at 3 amps, 250 volts.

GENERAL CONTROL—Discontinued Model MX-3, lightweight foot switch.

LITTELFUSE—Added No. 342008 at $.55 list to their 3AG fuse extractor post series.

MERIT TRANSFORMER—Added No. HVO-9 to their TV component series of horizontal output and hi-voltage transformers at $6.00 net.

PORCELAIN PRODUCTS—Introduced a series of porcelain lead-in tubing for UHF TV cable. This set comes in 7 different lengths from 3 feet to 12 feet, and prevents entry of rain or insects.

PREMIER METAL PRODUCTS—Introduced new series ARP, aluminum rack panels.

R.C.A.—Added TV component part No. 232T1, universal type horizontal deflection transformer (auto. trans. type) at $6.30 net.

RAM ELECTRONIC SALES—Decreased price on full focus deflection yoke No. Y70F30/3 to $6.00 net.

STANDARD TRANSFORMER—Added deflection yokes No. DY-1A at $4.74 net . . . No. DY-7A at $5.37 net . . . No. DY-8A at $6.60 net . . . No. DY-9A at $6.60 net and No. DY-10A at $6.60 net, all with leads and networks added.

Recording Equipment, Speakers, Amplifiers, Needles, Tape, Etc.

ASTATIC CORP.—Discontinued Model S-8, crystal pickup arm.

AUDIO DEVELOPMENT—Discontinued No. 111A, microphone cable transformer.

GENERAL ELECTRIC—Added No. RPX-052 at $38.95 list and No. RPX-053 at $57.90 list to their triple play variable reluctance cartridge series. Also added No. RPX-061 at $33.95 list and No. RPX-063 at $33.95 list to their variable reluctance cartridge series.

PACIFIC TRANSDUCER—Added Model 201D at $33.00 net. This model is the same as Model 201, a wide-range variable reluctance cartridge with a frequency velocity response to above 12,000 cps and with an output of 60 millivolts, with a diamond stylus instead of a standard sapphire stylus.

PERMOFLUX CORP.—Added No. DHS-31B at $50.00 net to their monaural dynamic headset line. Also added No. PHA-6 at $8.00 net and PHA-8 at $13.00 net to their line of headbands.

RECOTON—Discontinued their series of juke box osmium phono needles.

Test Equipment

HYTRON—Introduced No. SH27, test adapter at $1.45 net. The test adapter permits easier and faster "topside" chassis testing, since it can be plugged into chassis sockets with the replaced tube plugged into the top of the adapter.

JACKSON ELECTRICAL INSTR.—Added Model 710, selenium rectifier test set at $29.50 net. This pocket size unit tests all selenium rectifiers rated from 20 to 650 ma, and has a variable indicated voltage range from 25 to 300 volts a.c. It is designed to operate on 110 to 125 volts a.c. and includes "line adjust" control to give more accurate readings.

SIMPSON ELECTRIC—Increased price on Model 476, Mirroscope, to $197.00 net. This oscilloscope uses a 5-in. cathode ray tube mounted in a vertical position, with the image reflected from a high grade mirror mounted in the adjustable cover at the top of the cabinet, bringing the viewing surface near eye level when used on work benches of normal height. Linear sweep frequency is continuously adjustable in five overlapping ranges from 15 to 60,000 cps.

PROFESSIONAL DIRECTORY

Industry People...

Annual banquet of the Radio Club of America brought forth an assemblage of prominent old-timers long to be remembered—among those present and accounted for were **Paul Godley**, who sent the first Transalantic message; **Maj. E. H. Armstrong**, who gave us regeneration, the super-het, and FM; **Capt. J. H. Round**, who pioneered radio direction-finding while associated with Marconi; **Harry Houck**, World War I associate of Maj. Armstrong's, who today is president of Measurements Corporation; **Art Odgers**, president of Northern Radio Corporation; **Lincoln Walsh**, designer and manufacturer of the Brook amplifier—and so on far into the night.

Jerry Kahn, president of Stancor, and **Ken Prince**, director of Chicago's annual Electronic Parts Show, made rounds of friends while in Manhattan to conduct drawings for Parts Show exhibit space . . . **Maximilian Weil**, president, Audak Company, appears to be the industry's most consistent lander of publicity in national magazines—currently appearing for a one-month run in the January issue of "House Beautiful" . . . **F. M. Parsons**, vice-president, Kellogg Switchboard and Supply Company, Chicago, announces transfer of general sales offices to 79 W. Monroe St.

Robert W. McElroy has been promoted to newly-created position of director of safety and security for Minnesota Mining & Manufacturing Co. and subsidiaries . . . **Henry C. Roemer**, president, Federal Telephone and Radio Corporation, announces appointment of **Harold R. Hunkins** as chief engineer of company's Selenium-Intelin Division . . . **Wilbur G. Small** is new manager of exhibits for Raytheon Manufacturing Company . . . **Jack Berman**, vice-president in charge of sales for Shure Brothers, Inc., Chicago, for the past 13 years, is resigning to open his own sales rep firm in Southern California effective February first.

Max Baume, formerly sound department director, Hudson Radio & Television Corp., New York, is new sales manager for Brook Electronics, Inc., Elizabeth, N. J. . . . New sales and advertising manager for Drake Manufacturing Company, Chicago, is **Rex Munger**—appointment announced by **Kenneth Foute**, president . . . **Ralph Glover**, product manager of Jensen Manufacturing Company, Chicago, has been named a member of the RTMA's new High-Fidelity committee—he also heads RTMA Loudspeaker committee . . . **Paul V. Pembricks**, formerly head of the methods department, has been upped to assistant plant manager of Hammerlund Manufacturing Company . . . **Allen W. Walz** has been advanced by Arma Corporation to executive staff assistant to **Clifton T. Foss**, vice-president in charge of engineering.

Sidney Harmon, recently resigned as vice-president and general manager, David Bogen, Inc., New York, has acquired half interest in the firm of Harmon-Kardon, Inc., also New York, manufacturers of electronics equipment and audio-visual products—firm formerly was known as Kardon Manufacturing Corp. and was owned outright by **Bernard Kardon**, also a former vice-president and general manager of Bogen . . . **A. J. Nelson Company**, Denver, has been appointed sales representative for Permoflux Corporation in the Rocky Mountain area . . . **Mickey Freeman** has resigned as publicity director of KLAC-TV, Los Angeles, to fulfill similar duties for Hoffman Radio Corporation . . . **Raymond L. Kelley**, with Shure Brothers, Inc., Chicago, since 1946, has been appointed comptroller and vice-president in charge of finance.

Terminal Radio Corp., 85 Cortlandt St., New York, has been named exclusive area distributor of the Travis Tapak — new, self-powered, lightweight, portable tape recorder.

ADVERTISING INDEX

AT LAST!

no more unscheduled program breaks!

irish SOUND-PLATE

AN ENTIRELY NEW MAGNETIC RECORDING TAPE

It's break-proof...just what you've been wanting

(Produced and Sold Under the IRISH Label as 220RPA Red Band ORRadio Magnetic Tape)

No Other Tape Matches These Sound-Plate Features:

- **SOUND-PLATE** has two times the tensile strength of acetate
- **SOUND-PLATE** will not tear or break
- **SOUND-PLATE** eliminates timing problems —programs on the nose
- **SOUND-PLATE** is unaffected by moisture or temperature change
- **SOUND-PLATE** is engineered for high speed operation, 15 to 500" per sec
- **SOUND-PLATE** is not affected by static
- **SOUND-PLATE** has a greater surface slip

Another Reason WHY SOUND-PLATE is Best For Professional Use!

There are differences in magnetic oxides. ORRadio molecular lubricated oxides are more stable in coating conditions and turn out more uniform dispersions, thus assuring you of the finest recordings possible. SOUND-PLATE is made in the ORRadio tradition.

Here, at last, the sum total of everything you have ever wanted in a professional magnetic recording tape. ORRadio IRISH 220RPA SOUND-PLATE eliminates your physical problems in magnetic tape recordings. It won't tear or break. It has all the excellent qualities that have made ORRadio IRISH 211RPA the most popular tape among professional engineers.

Yes, SOUND-PLATE costs a little more . . . but, in the long run, it will prove to be not only the most satisfactory, but the most economical magnetic recording tape you can use.

Try SOUND-PLATE at our expense! Buy a reel of SOUND-PLATE at your favorite Radio Parts Distributor. If you are not completely satisfied, return it to your distributor who has been authorized to refund your money.

1200 Ft. SOUND-PLATE on Plastic Reel .... **$15.50 List**

2400 Ft. SOUND-PLATE on Metal Reel **$33.85**

SOUND RECORDING

REG U S PAT OFF

Manufactured in U.S.A. by

ORRADIO INDUSTRIES, INC. OPELIKA, ALABAMA

World's Largest Exclusive Magnetic Tape Manufacturer

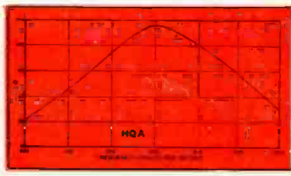
PERMALLOY DUST TOROIDS FOR MAXIMUM STABILITY...

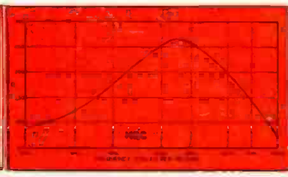
The UTC type HQ permalloy dust toroids are ideal for all audio, carrier and supersonic applications. HQA coils have Q over 100 at 5,000 cycles... HQB coils, Q over 200 at 4,000 cycles... HQC coils, Q over 200 at 30 KC... HQD coils, Q over 200 at 60 KC... HQE (miniature) coils, Q over 120 at 10 KC. The toroid dust core provides very low hum pickup... excellent stability with voltage change... negligible inductance change with temperature, etc. Precision adjusted to 1% tolerance. Hermetically sealed.

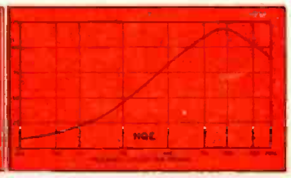
HQA, HQC, HQD CASE
1 13/16" Dia. x 1 3/16" High

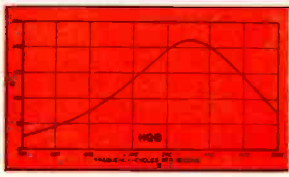
HQB CASE
1 5/8" x 2 5/8" x 2 1/2" High

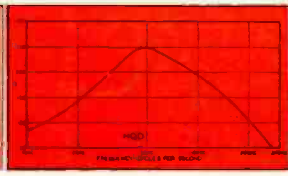
HQE CASE
1/2" x 1 5/16" x 1 3/16" High

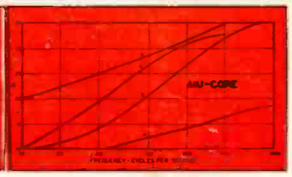








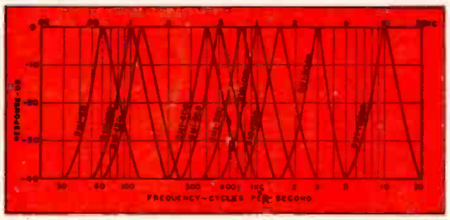


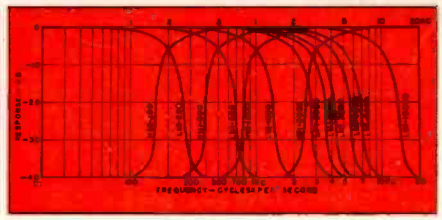


Type No.	Inductance Value	Net Price	Type No.	Inductance Value	Net Price	Type No.	Inductance Value	Net Price
HQA-1	5 mhy.	$7.00	HQA-16	7.5 hy.	$15.00	HQC-1	1 mhy.	$13.00
HQA-2	12.5 mhy.	7.00	HQA-17	10. hy.	16.00	HQC-2	2.5 mhy.	13.00
HQA-3	20 mhy.	7.50	HQA-18	15. hy.	17.00	HQC-3	5 mhy.	13.00
HQA-4	30 mhy.	7.50	HQB-1	10 mhy.	16.00	HQC-4	10 mhy.	13.00
HQA-5	50 mhy.	8.00	HQB-2	30 mhy.	16.00	HQC-5	20 mhy.	13.00
HQA-6	80 mhy.	8.00	HQB-3	70 mhy.	16.00	HQD-1	.4 mhy.	15.00
HQA-7	125 mhy.	9.00	HQB-4	120 mhy.	17.00	HQD-2	1 mhy.	15.00
HQA-8	200 mhy.	9.00	HQB-5	.5 hy.	17.00	HQD-3	2.5 mhy.	15.00
HQA-9	300 mhy.	10.00	HQB-6	1. hy.	18.00	HQD-4	5 mhy.	15.00
HQA-10	.5 hy.	10.00	HQB-7	2. hy.	19.00	HQD-5	15 mhy.	15.00
HQA-11	.75 hy.	10.00	HQB-8	3.5 hy.	20.00	HQE-1	5 mhy.	6.00
HQA-12	1.25 hy.	11.00	HQB-9	7.5 hy.	21.00	HQE-2	10 mhy.	6.00
HQA-13	2. hy.	11.00	HQB-10	12. hy.	22.00	HQE-3	50 mhy.	7.00
HQA-14	3. hy.	13.00	HQB-11	18. hy.	23.00	HQE-4	100 mhy.	7.50
HQA-15	5. hy.	14.00	HQB-12	25. hy.	24.00	HQE-5	200 mhy.	8.00

UTC INTERSTAGE AND LINE FILTERS

FILTER CASE M
1 3/16 x 1 11/16,
1 5/8 x 2 1/2 High





These U.T.C. stock units take care of most common filter applications. The interstage filters, BMI (band pass), HMI (high pass), and LMI (low pass), have a nominal impedance at 10,000 ohms. The line filters, BML (band pass), HML (high pass), and LML (low pass), are intended for use in 500/600 ohm circuits. All units are shielded for low pickup (150 mv/gauss) and are hermetically sealed.

STOCK FREQUENCIES
(Number after letters is frequency)
Net Price $25.00

BMI-60	BMI-1500	LMI-200	BML-400
BMI-100	BMI-3000	LMI-500	BML-1000
BMI-120	BMI-10000	LMI-1000	HML-200
BMI-400	HMI-200	LMI-2000	HML-500
BMI-500	HMI-500	LMI-3000	LML-1000
BMI-750	HMI-1000	LMI-5000	LML-2500
BMI-1000	HMI-3000	LMI-10000	LML-4000
			LML-12000

United Transformer Co.

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EXPORT DIVISION: 13 EAST 40th STREET, NEW YORK 16, N. Y., CABLES: "ARLAB"