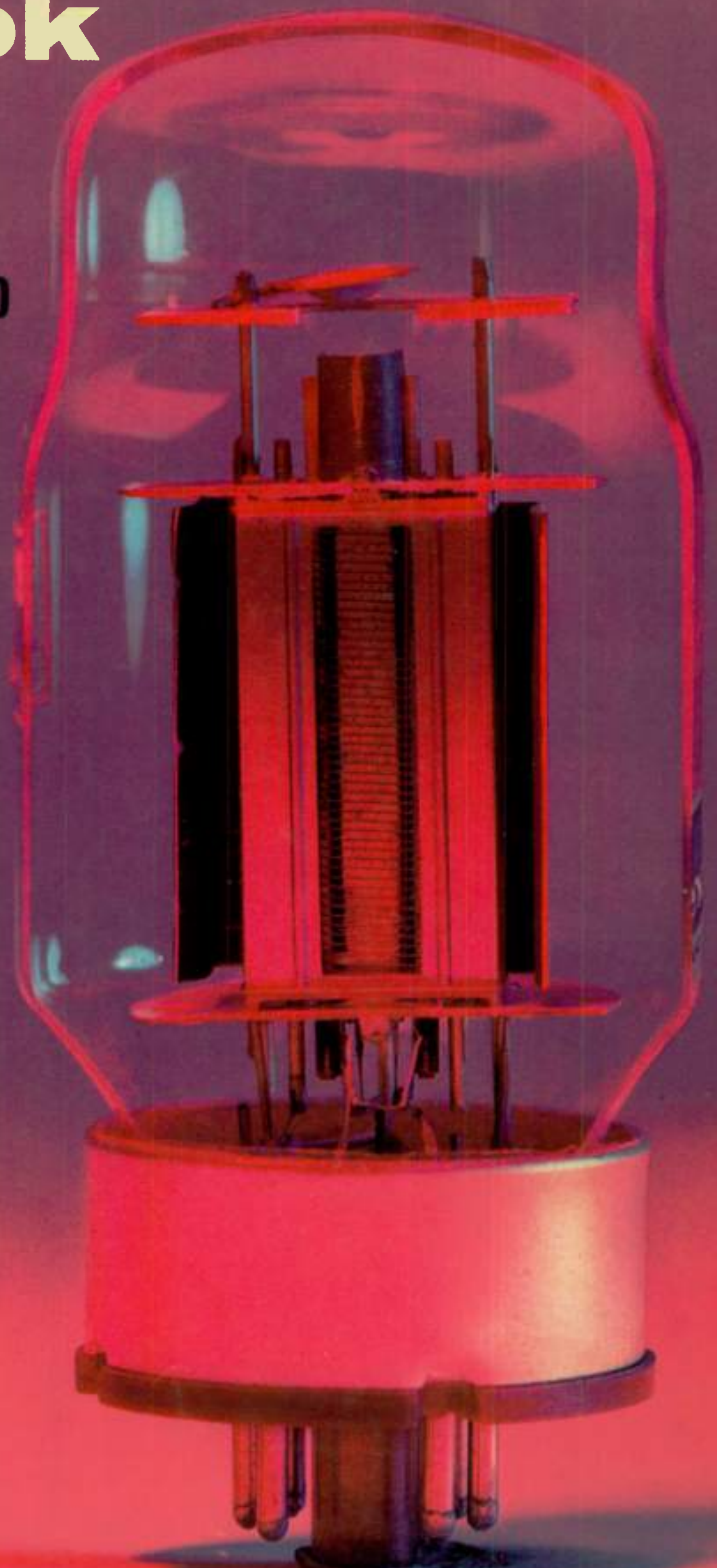
AN ELECTRONICS WORLD ANNUAL

audio yearbook 1961

$1.00 (1.25 in Canada)



ROOM ACOUSTICS FOR STEREO
Special 4-Part Section

BUILD: A Compact 20 Watt Amp • Simple FM Tuner Booster • Low Noise Transistor Preamp

Check Your Hi-Fi System For 20 Common Ailments

Power Output Rating For Hi-Fi Amplifiers

PLUS: More Than 20 Features On Better Audio


They Reign Supreme...

Announcing a recording tape superior enough to be called Ampex

The same exacting engineering skills that made Ampex the leader in the tape recording field have now created the world's *finest* recording tape! Only Ampex could have engineered a tape with so many superior performance characteristics . . . a tape that truly merits the name Ampex. Here are some of the reasons: *greater dynamic range, 4 to 8 db wider than any other tape; lower print-through, 6 to 8 db lower than any other tape.*

Even the package is unique! Ampex SIGNATURE-BINDING, enclosed with every reel, lets you personalize your recordings with distinctive gold lettering on maroon and green leatherette bindings.

Irish Recording Tape, too, is a quality Ampex product...surpassed *only* by Ampex Recording Tape, the world's finest! Hear the proof now at your Ampex Tape Dealer's.

AMPEX MAGNETIC TAPE PRODUCTS / ORR INDUSTRIES COMPANY
A Division of The Ampex Corporation, Opelika, Alabama

audio yearbook

1961

Oliver Read, Publisher

Thomas S. Kneitel, Managing Editor

Rodney Marquardt, Art Director

Fred U. Wetzler, Associate Editor

Barbara R. Brennan, Assistant Editor

John A. Ronan Jr., Advertising Director

George E. Morrissey, Advertising Manager

OUR COVER shows a cutaway view of the famous KT-66 beam tetrode tube, the heart of many a fine stereo and mono system. The KT-66, by the way, was selected by D. T. N. Williamson for his original amplifier and has since been used in many other circuits. It is manufactured by the General Electric Company of England (Genalex). *Photo by Albert Gruen.*

ZIFF-DAVIS PUBLISHING CO., One Park Ave., New York 16, N. Y.
William B. Ziff, Chairman of the Board (1946-1953);
William Ziff, President; W. Bradford Briggs, Executive Vice President;
Michael Michaelson, Vice President and Circulation Director;
Hershel B. Sarbin, Vice President; Charles Housman, Financial Vice President;
J. Leonard O'Donnell, Treasurer.

CONTENTS

Section 1:

Stereo and Stereo Speaker Systems

The Stereo Illusion 6
More About Wide-Stage Stereo 12
Tone Arm Tracking Error in Stereo 18
Stereo Microphone Techniques 23
Subdivided Hi-Fi Speaker Enclosure 26
The Bi-Phonic Coupler 30
Extended Stereo System 34

Section 2:

Hi-Fi and Audio Theory and Practice

Finding Faults In Hi-Fi Systems 38
Simplified Speaker Testing 41
Intermodulation Distortion 45
The Decibel Without Pain 48

Section 3:

Amplifiers, Tuners, Phonos, Tape

Power Output Ratings For Hi-Fi Amplifiers 54
Compact 20-Watt Hi-Fi Amplifier 57
Bass-Treble Hi-Fi Amplifier 60
Low Noise Transistor Preamp 65
Tape System For Stereo 68
Transistorized FM Multiplex Adapter 71
Multiplexing Music With One Recorder 74
New AM-FM Tuner Standard 78
Simple Booster For FM Tuners 81
A Sweep Generator For Hi-Fi AM 84

Section 4:

Room Acoustics For Stereo

Part 1 – Basic Principles 90
Part 2 – Reverberation and Room Treatment 94
Part 3 – Sound Absorbers and Special Speakers 99
Part 4 – Balancing The Room 104

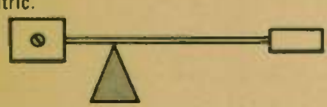
Now the Garrard Laboratories Proudly Announce an Entirely New Kind of Record Playing Unit

AN AUTOMATIC TURNTABLE

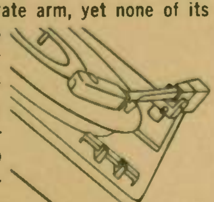
GARRARD'S LABORATORY SERIES TYPE A

Now, at last, you can enjoy all the advantages of a true, dynamically-balanced tone-arm (with a built-in calibrated pressure gauge), a full-size, heavy-weight professional turntable, a laboratory-balanced precision motor...plus the much-wanted convenience of the world's finest automatic record-player ...all in one superb instrument! △ No one but the Garrard Laboratories, with their unmatched facilities, could have accomplished it. With 40 years of manufacturing experience, and the highest engineering and precision standards in the Industry, Garrard set out to develop an all-in-one unit that would satisfy every critical requirement, even surpassing the professional turntable standards established by the NARTB.

THE ONLY DYNAMICALLY-BALANCED TONE ARM ON AN AUTOMATIC UNIT Now, for the first time, a tone arm which meets the very latest engineering standards established by the Industry—has been incorporated into an integrated record-playing unit. This highly advanced tone arm is put in perfect dynamic balance by moving an adjustable counterweight. At this point, it is in "gyroscopic" balance, with zero pressure. To set the tracking pressure designated for any cartridge, a pointer is moved along a calibrated scale at the side of the arm which is graduated in grams. This built-in stylus pressure gauge now shows the precise tracking force. The accuracy of this setting is even greater than that which could be measured by any separate stylus pressure gauge. The arm will now track correctly even if the player is intentionally tilted, or if the record is warped or not perfectly concentric.



And incidentally, regardless of the number of records on the turntable, the angle at which the stylus meets the record is negligible, due to the unique geometry of this arm. △ Since all of these engineering requirements guarantee that there is no unequal pressure on the sides of the stereo record grooves... distortion, channel imbalance, record and stylus wear are eliminated, resulting in perfect stereo reproduction. But "perfect performance" also requires minimum friction, and this is assured by the two precision needle pivots on which the arm is set. This arm is precision-mounted for you, thus affording all the advantages of the separate arm, yet none of its inherent disadvantages. The danger of the tracking error which often occurs in the mounting of separate tone arms, is eliminated completely because there is no possibility of even the slightest mislocation of the arm.



FULL-SIZED, HEAVILY WEIGHTED, BALANCED, CAST, POLISHED TURNTABLE The weight of six pounds has been determined as the optimum for perfect balance, torque and flywheel action in this unit. This combination insures silent, on-speed running. The Garrard engineers have conceived of this new turntable as a "sandwich". It is actually two turntables balanced together...a drive table inside, and a heavy, cast turntable outside. The two turntables are separated by a resilient foam barrier, which effectively damps out any possibility of noise or vibration. Furthermore, being of non-ferrous metal, the cast tables offer no attraction to sensitive magnetic pickups.

NEW "LABORATORY SERIES" MOTOR The Type A is built around a newly-developed Garrard four-pole shaded motor, which was designed especially for it. This laboratory motor is the perfect match for the new turntable system, and provides quiet accuracy, regardless of load or voltage changes. Constancy of speed is such that this motor will bear the closest scrutiny by sensitive measuring instruments. The armature is micro-balanced on exclusive Garrard equipment, and free of vibration. The total result is true musical pitch and clear sustained musical passages, without the irritation of wow or flutter. The "Laboratory Series" motor is completely shielded, top and bottom, with specially-designed and accurately oriented plates, insuring the absence of hum, even with the most sensitive magnetic pickups. To minimize even the slightest vibrations, the entire motor assembly is isolated from the unit by shock mounts of a special formula and design.

THE GREAT PLUS FEATURE OF AUTOMATIC PLAY—WITHOUT COMPROMISE! The convenience and desirability of being able to play records automatically, and to have the unit shut itself off, have long been recognized and accepted. Certainly, even the most critical user would want these features, but only if they could be incorporated without compromising the quality, performance or inducing record wear. This has now been accomplished in the Type A, by adding Garrard's exclusive pusher platform changing mechanism, with its smooth, one-piece center spindle, to the arm, turntable and motor described above. This unique combination of features means that for the first time ever, there is a record playing unit which answers every requirement of both performance and convenience. It accomplishes this to such a degree that it is certain to appeal to every person, regardless of whether he already owns equipment, or is planning to buy a new system. The fact is that the Type A obsoletes all previous equipment. It protects records better. It has performance and features superior to separate turntables and arms. It also incorporates professional characteristics not found in any record changers.

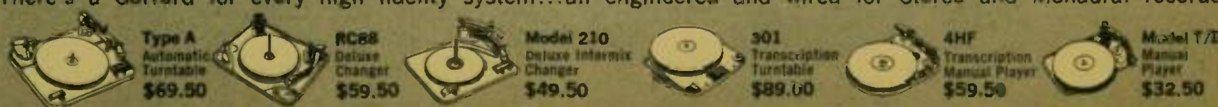
The concept of this great new Garrard record-playing unit was extravagant, but the price of **$69.50** is surprisingly modest—made possible only because of Garrard's extensive facilities, highly developed production methods and critical quality control procedures. △ If there ever was a single record-playing device which answered every requirement of every music system—we believe this is it. △ And with the Type A, once and for all let us lay the ghost that simply because a record player makes available certain automatic conveniences (which you may or may not use at your option)—that this in any way implies that the unit cannot be actually superior to a separate turntable and arm. △ For in this bold new unit you will find the realization of everything you have wanted in a record player. Examine it thoroughly, and you'll want to install the Garrard Laboratory Series Type A in your own music system—now. △ Your dealer has—or soon will receive—the LABORATORY SERIES TYPE A GARRARD. Let your own eyes and your own searching examination, prove beyond doubt that this is indeed the record player for you. Or, if you prefer, write today for illustrated, descriptive booklet.

Garrard

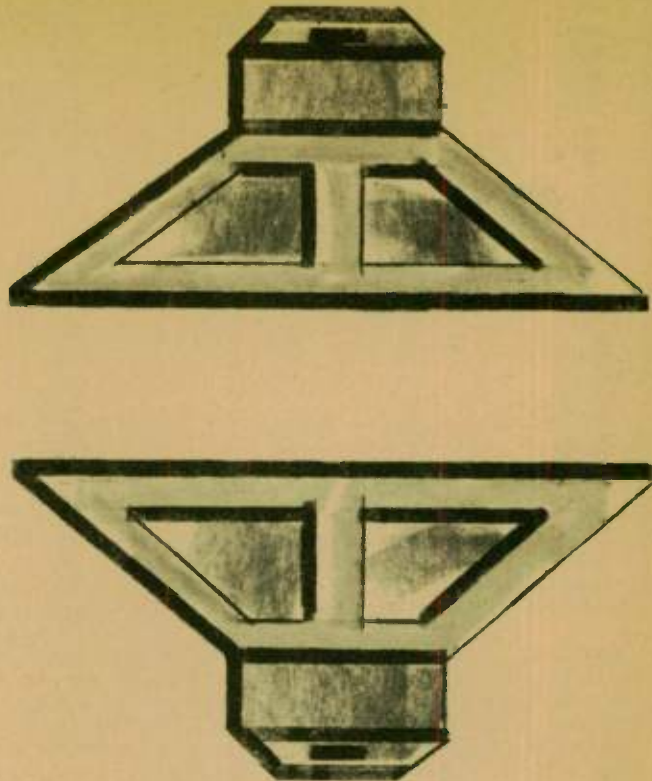
Write Department GI-1170
Garrard Division of British Industries Corporation, Port Washington, New York.

B I C ENDORSED QUALITY

There's a Garrard for every high fidelity system...all engineered and wired for Stereo and Monaural records.



Canadian inquiries to Chas. W. Pointon, Ltd., 66 Racine Road, Rexdale, Ontario Territories other than U.S.A. and Canada to Garrard Engineering & Mfg. Co., Ltd., Swindon, Wilts., England



SECTION·1

Stereo and Stereo Speaker Systems

The Stereo Illusion	6
More About Wide-Stage Stereo	12
Tone Arm Tracking Error in Stereo	18
Stereo Microphone Techniques	23
Subdivided Hi-Fi Speaker Enclosure	26
The Bi-Phonic Coupler	30
Extended Stereo System	34

THE STEREO ILLUSION

By **J. DONALD HARRIS,** Ph.D.
U. S. Naval Medical Research Laboratory

A psycho-acoustician examines stereo and explains how the illusion is re-created for the listener.

OUR AUDITORY WORLD is an amazingly complex series of concentric spheres. The head occupies the center and seems, for the most part, to be stationary while sound emitters shift and recede and crisscross around us (Fig. 1). The problem of how to locate and track a single moving sound source, or even several simultaneous sound sources, has been met and solved by nature in a variety of ways. The insect, the bat, the porpoise, and man have each worked out special compromises—some more successful than others.

The chief function of the ears of primitive vertebrates was certainly direction-finding. To promote this function the ear's frequency sensitivity had to go far beyond the low-frequency hearing of reptiles, and evolve "directional antennas" and the muscles for directing these "antennas" both together and, to some extent, independently. This improved direction-finding is at the expense of time so that the eye, at least in the primate, evolved simultaneously into an instrument with an elaborate oculo-motor system for target tracking. We thus have, in the human, the equivalent of an omnidirectional search radar (the ear) and the extremely directional fire-control radar (the eye). See Fig. 2.

Once our ears have put our eyes "on target" they can pretty well be dispensed with and, in fact, the high-frequency-range capabilities and the large mobile outer ear parts of the lower animals are disappearing rather rapidly—as such things go. We no longer boast the exquisite hearing in the higher octaves—enjoyed by the dog and the cat—nor are we independent of our eyes for ordinary locomotion.

Despite these evolutionary changes, a great deal of "liveness" and three-dimensionality exist in our auditory world. Much has been done in recent years to exploit these advantages. Applications have even outstripped theory in this case since the stereophonic en-

gineers have accomplished remarkable things with multiple microphones and speaker arrays. On the other hand, scientists have had to content themselves with a single stationary sound source and have even had to resort to the artificiality of earphone listening. To date no psycho-acoustic laboratory has developed the essential array of moving multiple sound sources in a large anechoic chamber nor devised the required servo systems and X-Y recorders to program and record the stimuli and the subject's response. One can only anticipate such reasearch for the next decade or two.

Nevertheless, a unique and even exciting chapter in psycho-acoustics embraces a number of interesting discoveries regarding the directionality of human hearing, the relative importance of the several possible "cues" for directionality, the conditions under which directionally may be improved, and mathematical-statistical models of binaural hearing.

Two Ears Better than One?

The first question that must be resolved is whether two ears are really better than one. The answer, for all practical purposes, is *negative* for almost all aspects of hearing except

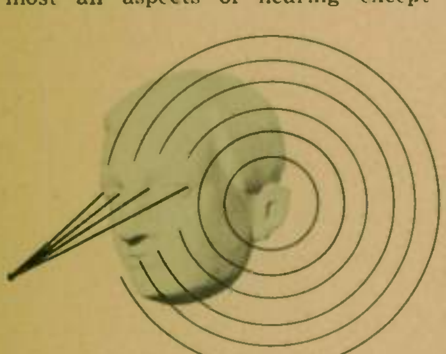


Fig. 2. Our ears serve as search radar, our eyes as fire-control radar.

directionality. For contact detection or sensory discrimination any improvement of the binaural combination over the monaural involves only a decibel or two—a negligible percentage. For localization of sounds in space, however, the monaural mode has almost nothing to recommend it. Whether one is judging distance to a sound source or angular direction, a second ear is almost indispensable. Probably the best confirmation for this statement involves tests made with bats. While normal bats will be able to avoid wires strung around a room up to 76% of the time, bats with one ear covered, however, have an avoidance record of 38-41%—35%, in this case, being due to chance alone.

One asks, then, how does this come about? Both ears hear the same thing so how can the same information, being 100% redundant, fed into the second ear, add what amounts to *new* information on the basis of which we can make further and remarkably precise judgments?.

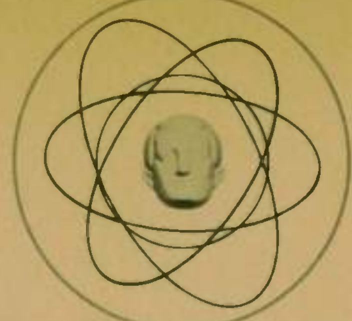


Fig. 1. We all live in an auditory world made up of concentric spheres.

The secret is that when one takes not a first but second look at the information reaching the two eardrums at any moment in time, the acoustic signals can usually be shown to be markedly different in their "hyperfine" details.

The temporal (or time determining) characteristics of the ear, its frequency and intensity handling capacities, and the layout of its nervous system are admirably adapted to make a distinction on the basis of just those hyperfine details by which the two acoustic signals differ. By "make a distinction", we mean that the mechanical construction of the ear itself does not blur any of the features of the acoustic signal but passes them on, transformed, to the auditory nerve where they are again transformed (neural coding), sent upward to the brain, and decoded, with surpassing exactitude, into sensation.

Temporal Characteristics

Let us first consider the temporal characteristics (time of arrival and phase) of the peripheral organ of hearing. If one presents a series of clicks to the ear, they do not merge into a continuous tone until the repetition rate reaches many hundreds per second. Contrast this phenomenon with the eye, where the critical flicker frequency is something under 50 per second—a relatively slow-acting unit. But the ear is not a biochemical system with slow time-constants like the eye—it is a mechano-hydro-electrical system with almost critical damping. (It is of course true that several distingishable d.c. and a.c components within the cochlea—or inner ear—are maintained by bio-chemical-metabolic equations, but the stimulation of the auditory nerve fibers is true electrical stimulation, generated by the hair cells within the inner ear.)

Only by keeping in mind the very fast-acting nature of the peripheral organ of hearing can one believe the almost incredible performance revealed by laboratory tests. If one keeps all other physical parameters constant except the time of arrival at the eardrums of a short burst of noise, the subject can detect a difference in arrival time of as little as *6 millionths of a second*. Evidently the end-organ and its associated nervous system carry and transform time patterns with great fidelity—certainly with sufficient fidelity to provide an accurate sense of direction on the basis of time of arrival of the stimulus at the two ears.

Let us assume, for easy calculation, that sound travels one foot per millisecond. In .00001 second it will travel .01 foot. In this case a sound source would have to be off the mid line by only a few degrees for one to sense, by the time-of-arrival cue alone, that the sound was not, in fact, in the median plane. (See Fig. 3.) This cue can apply, of course, only to discontinuous sounds.

Note, however, that with a continuous pure tone another correlated cue is present, namely, time of arrival at the eardrums of a particular point in the phase. In this case the ears are sensitive to differences in phase at the eardrums at any moment in time. If one varies the phase (or inserts a time delay, which is the same thing) in one arm of a pure tone which is split and led to the two drums, one has the illusion of the sound moving back and forth between the two ears. Judgments of the (illusory) "sidedness", in degrees off the mid line of the tone, correspond surprisingly well to the actual geometry of a point source and an imaginary line between the ears where the geometry would, under actual acoustic conditions, produce the phase difference generated electronically. A lead or lag by 100 microseconds of a peak at one ear usually permits a clear judgment of "off mid line". (See Fig. 4.)

Phase relations as a localization cue in nature have, of course, a serious limitation where the distance between the two ears equals one wavelength. Here the peak-to-peak times of arrival are identical and non-informative. This is more apt to occur at high frequencies and with small heads. This is why the bat has abandoned continuous tones and emits, instead, a series of heavily frequency-modulated and extremely brief tones, thus utilizing frequency differences and times of arrival rather than phase for localization purposes.

Let us now consider the intensity handling capacities of the ear and the possible intensity cues to localization.

Of course, if the head is turned away from a sound source there will be a difference in intensity depending on the sound decay over the distance, but the cue is not a simple one even with the immobile pinna (outer ear parts)

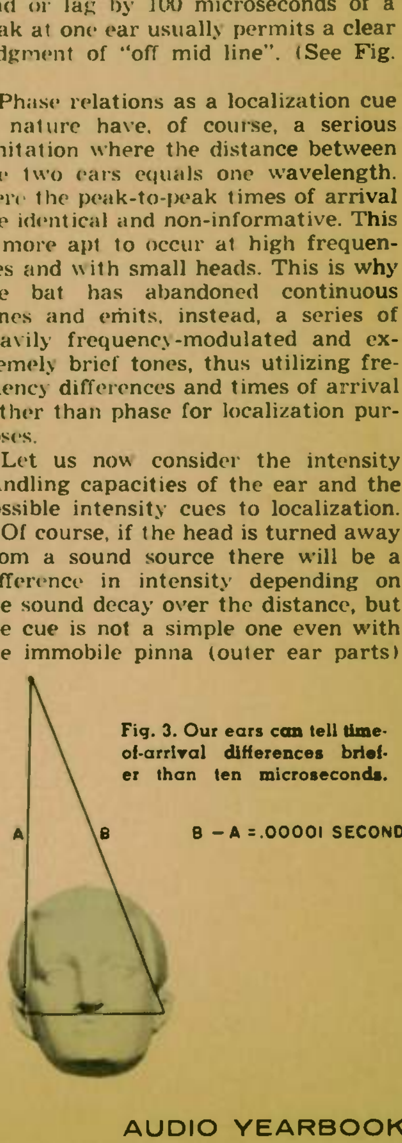
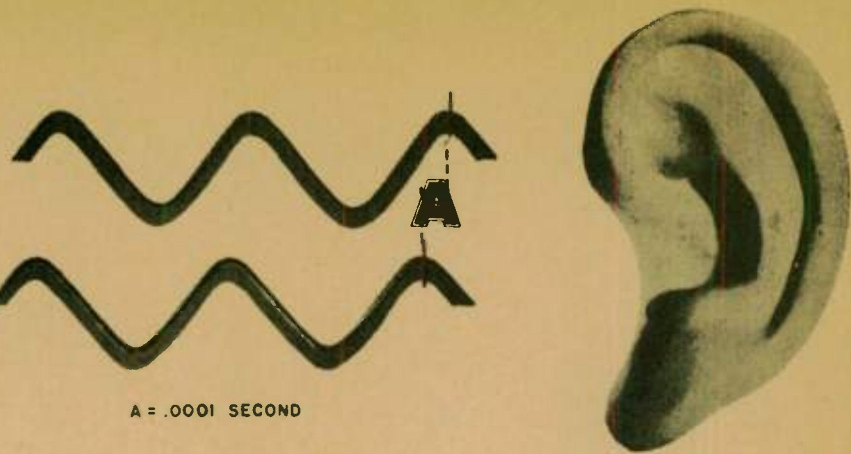


Fig. 3. Our ears can tell time-of-arrival differences briefer than ten microseconds.

Fig. 4. The human ear is very sensitive to phase differences below about 1500 cps. A lead or lag by 100 μsec. of a peak at one ear usually permits a clear judgment of "off mid line" mode.



of the human. The lower frequencies tend to bend around the head so that the sound "shadow" cast by the head introduces a relative bass-boost to the spectrum reaching the farther ear. (See Fig. 5.) Also, there are reinforcements and interferences brought into play by the configuration of the head and pinna. Note that the sensitivity of each ear *by itself* varies with the direction of the sound. The binaural condition must compensate for these differences. In animals with moving pinnae, these variations are probably magnified quite deliberately and used to provide further information.

A quick calculation will show that the intensity differences arising from movements of the head (or, what amounts to the same thing—moving the source off mid line) are much greater through many octaves than the differential loudness discrimination available to the average human ear. An 800-cps tone from one side casts an 8 db "shadow" on the opposite ear, while errors of mid-line localization can be quite negligible on the basis of the 0.5 db discrimination possessed by many individuals.

It is necessary to emphasize what we have stated previously that two distinct mechanisms underlie our sense of directionality—and these in different frequency regions. A graph showing the minimum audible angle, in degrees, to which the average head is sensitive in the median plane reveals a broad minimum at about 1 degree between 250 and 1000 cps and a rapid rise above 1000 cps to a maximum of about 3 degrees at approximately 1500 cps. For these lower frequencies, temporal characteristics (time of arrival, phase) underlie performance. Above 1500 cps, the minimum audible angle in the median plane fluctuates around 2 to 3 degrees up to 10,000 cps. For these higher frequencies the localization cue is provided by intensity differences (the sound "shadow").

When the minimum audible angle is measured for sound arriving at the head at an angle 45 degrees off the mid line, performance is generally similar to the mid-line condition but 1 to 2 degrees worse, up to about 1500 cps. But for higher frequencies, where intensity cues come into play, performance deteriorates so sharply that a very inefficient minimum audible angle of 10 degrees exists at 3000-4000 cps while directionality is, for all practical purposes, completely lost at 7000 cps and above.

Evidently the relations among head size, head shape, and wavelength are not such as to make intensity cues very useful in anything but the symmetrical mid-line condition. This fact must be considered when placing microphones and speakers to create the stereo illusion.

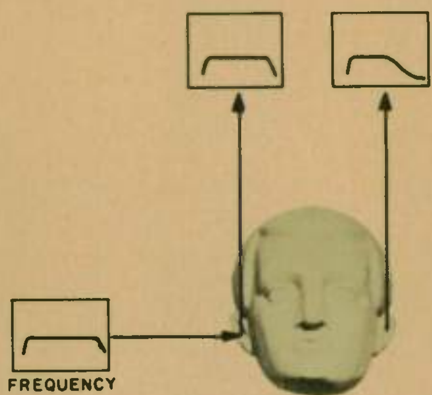
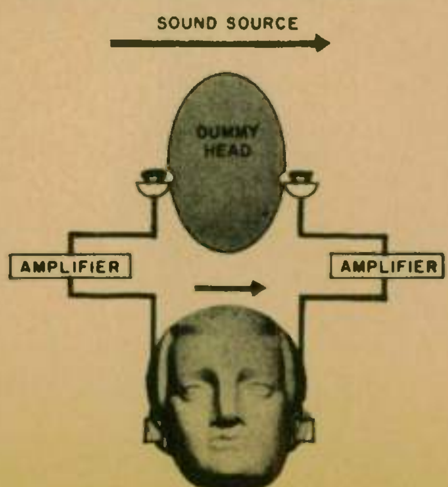


Fig. 5. The head casts a sound shadow that attenuates the higher frequencies.

Fig. 6. The stereo illusion in pure form employs two microphones in a dummy head.



To provide the strongest possible stereo illusion it is only necessary to re-create in a pair of ears, by any convenient means, the time and intensity conditions generated at a point in space by an actual sound source.

Re-creating the Illusion

Probably the most compelling illusion of a wide-azimuth sound, or of a moving sound source, is created by placing two omni-directional microphones in the simulated ear canals of a dummy head and recording the output on dual-channel tapes. This program material is then fed to two earphones. (See Fig. 6.) Here all cues heard by the ultimate listener are as if his head were in the position occupied by the dummy head and the illusion is complete.

If playback is to be through loudspeakers, other considerations are involved. The original recording studio and the listening room must exhibit certain basic similarities. The speakers must not be too far off mid line in relation to the listener. They must be matched closely in phase and the two channels must be symmetrical to within a few microseconds in time delay.

Fortunately, these rather stringent conditions need not be met 100 per-cent in order to create a satisfactory illusion. A considerable blurring can be tolerated without destroying the illusion and, of course, the two ears will always appreciate the added "liveness" and volume provided by the second channel. But it is very easy to mistake "liveness" and volume for true stereo illusion—which once heard in all its purity is an unforgettable experience.

This blurring can be studied in the anechoic chamber. If one places two microphones four inches apart then records a wide-azimuth sound, when replayed through a two-speaker array there will be a cube (about one foot on a side) where one can "immerse" the head and experience the stereo illusion. If the microphones are placed about a foot apart, the illusion volume will be a cube about a yard on a side. However, the illusion on the fringes of the cube loses some of its force—rather soon one encounters diminishing returns and the reasons are not hard to

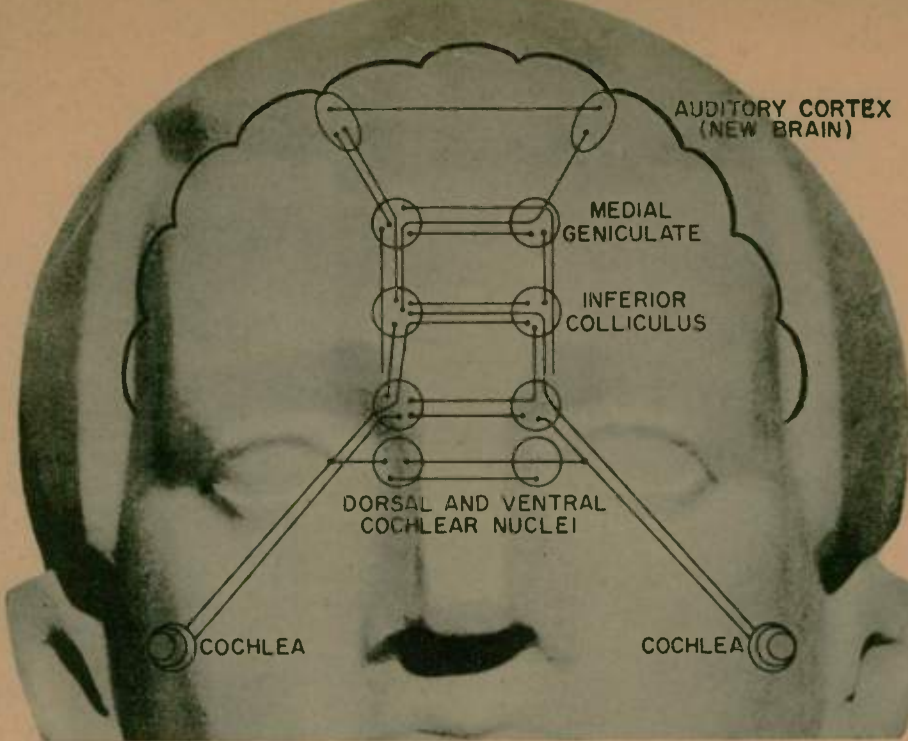


Fig. 7. Some connections in the ascending pathways of the auditory nervous system.

pinpoint. If the phase relations for tones 1500 cps and below, which finally reach the ear by way of the speakers, are not at all those which the ears would hear if immersed in the original sound, there is no reason to expect to re-create the exact illusion. The most one could expect would be a "diffuseness" to the sound so that it might be hard to say, if blindfolded, just where the sound source is in the room. And indeed this is about all that some systems and recordings achieve. It is not necessary, however, to be satisfied with this extension of simple monaural principles.

To the auditory scientist, of course, the compelling question is not how one shall create the stereophonic illusion but how is it possible that the binaural auditory system can be so extremely sensitive to temporal and intensive patterns. After all, the speed of the nervous impulse and delays in neurone-to-neurone connections are on the order of milliseconds—how then can the ears be sensitive to differences on the order of microseconds?

Auditory Nervous System

The answer to this question seems to lie in the layout of the auditory nervous system. (See Fig. 7.) The inner ear, though coiled, can be considered unrolled as a piano harp with a regular progression from the high frequencies to the low. The hair cells in this progression attach to and "trigger" the individual auditory nerve fibers. This impulse travels by several "relay stages" to the cortex, or new brain. Unlike a telephone wire through a switchboard, these fibers do not go straight to the cortex. They emerge from the cochlea and go a few millimeters to the brain stem where each nerve fiber splits and sends one fibril to the dorsal and one to the ventral cochlear nucleus. At this point some of the fibers terminate on another fiber (the juncture, called a "synapse," introduces a time delay) whereupon that fiber goes up the brain stem. On the other hand, some of the fibers from the cochlea continue straight through the dorsal and ventral nuclei with no synaptic delay and proceed up the brain stem.

Note that we are by no means up to the cortex at this point. It is very important to understand that some fibers at the level of the first nucleus cross over to the other side so that even at the first "relay" station we have the possibility of inter-aural effects.

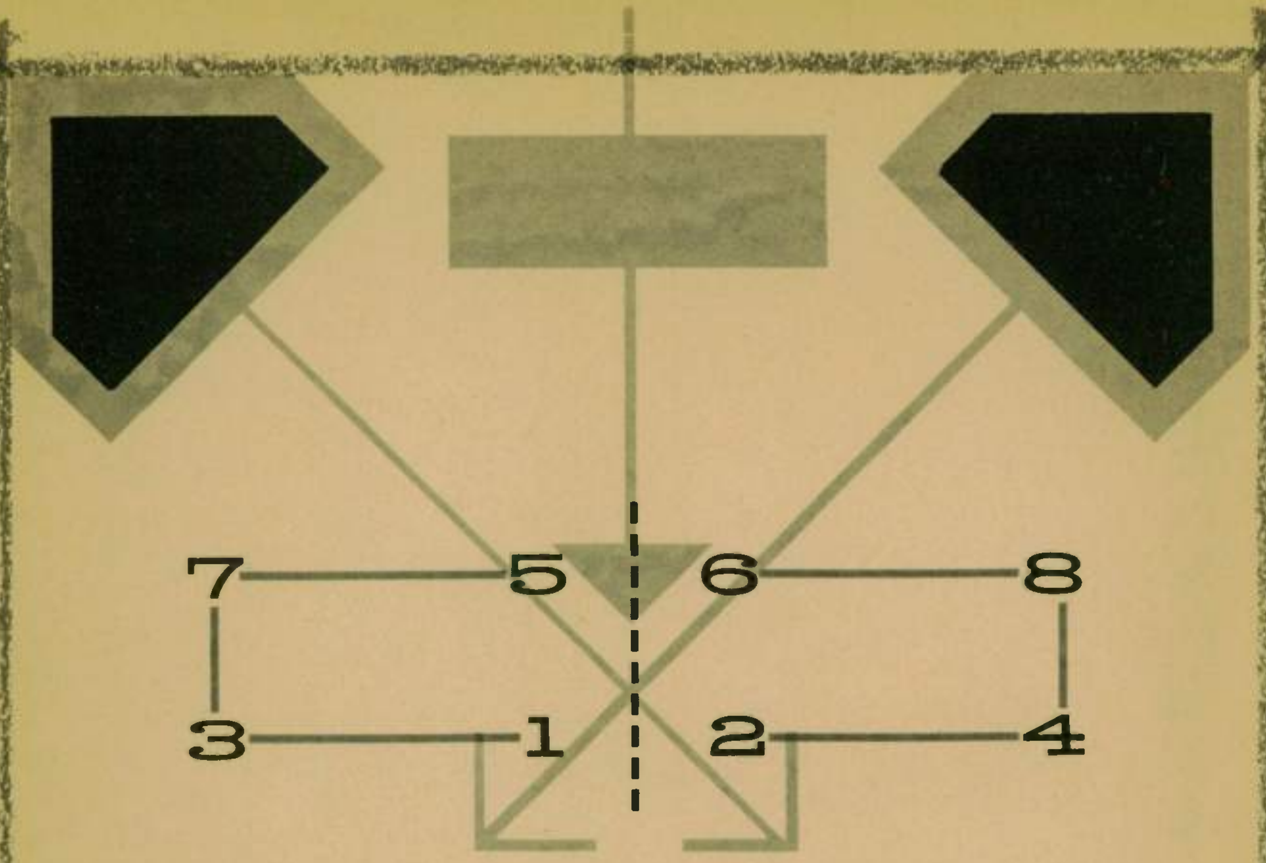
Now the crossed and uncrossed fibers, some synaptically delayed and some not, go up the brain stem to another left-right pair of nuclei called the "colliculi." Here, again, in each colliculus we have three possibilities. A "relay" can cross from the other side, or can go to the other side, and the colliculus can be bypassed by a fiber going straight through to the geniculate body in the mid-brain, just below the cortex.

What we have here is an ideal system for a comparison of stimuli impinging on the two ears. The opportunity for comparing the inputs to the two ears is good—so much so that the neural representation from the left ear is actually greater on the *right* side of the nervous system. There are not just a few fibers which cross. Furthermore the system is not analogous to the visual nervous system where 50 per-cent of the fibers also cross. There, the retina is divided into left and right halves, the fibers from the nasal half of each eye crossing in the optic chiasma—the fibers from the temporal half going back, uncrossed, to the visual cortex. But this is a simple and uncomplicated layout compared to the "relay" systems, "delay lines," and "switchboard" opportunities in hearing.

For the mathematically minded, the ladder system as we see it here allows for two types of correlation. First a running auto-correlation of one ear on itself. Built-in "delay lines" at the synapses and arising from different fiber lengths make this possible—from a single half-millisecond click we may have nerve impulses reaching a "switchboard" over a period of 5, 10, or 20 milliseconds. Thus a multiple look can be had, within each of several successive "switchboards," at a single monaural input.

How much more, then, is the opportunity laid down for a second type of correlation—a running correlation between the two ears. At every "station" from the brain stem to the cortex a multiplex and indeed superb look can be had of the two inputs simultaneously. It is in the course of these repeated observations that the nervous system shapes and refines the binaural sensation.

Opening a "second ear" to the world results in a dramatic sharpening of auditory experience and in worthwhile improvement in orientation and naturalness. We have seen how this is made possible on the basis of time and intensity cues, with a nervous system designed, as if by a superb architect, for the full utilization of those cues. Future electro-acoustic systems will undoubtedly exploit to the full these capabilities of our binaural sense.



More About Wide-Stage Stereo

By **PAUL W. KLIPSCH** / Klipsch and Associates, Inc.

For stereo coverage of a wide listening area the addition of a center channel speaker along with a pair of corner speaker systems is recommended.

STEREO ENHANCES reproduced sound by supplying the sensations of depth, improved definition, and enlargement of apparent volume of the listening room.

Defining "high-fidelity" as the accurate reproduction of original tonality has its counterpart in stereo as the accurate reproduction of the geometry of the original sound. The two should convey to the listener the mental picture of the original sound, both in tonality and geometry. The essentials of stereo are:

1. Breadth or apparent width of the sound source.
2. Spatial continuity or a continuum of sound rather than several point sources.
3. Directionality or the ability of the observer to locate sounds across the stage in approximately the locations as originally generated.

Steinberg and Snow[1] showed that three sound channels were necessary and sufficient to achieve a reasonable approach to these requirements. Experiments by this writer confirm their conclusions, including the feasibility of deriving three channels from two sound tracks.

Two-Speaker Stereo

The author's own early experiments in stereo led to the conclusion that if two speakers were far enough apart to produce a satisfactory stage width, there was a tendency toward a two-source effect.

In a 16 foot by 25 foot room, the various arrangements of speakers shown in Fig. 1 were tried. In Fig. 1A, the stereo effect was noted only at close proximity to the array; in Figs. 1B and 1C, the listening distance was greater but the two-source or "hole-in-the-middle" effect was objectionable. In Fig. 1D there was a lack of stereo effect while in Fig. 1E, the stereo effect was evident, but the dependence on wall reflections resulted in poor mid- and high-frequency response and a hole-in-the-middle. In all cases, focus of a soloist was possible only for one

Here's your happy hi-fi hunting ground...

THE STROMBERG-CARLSON CATALOG OF THE FINEST IN HIGH-FIDELITY AND STEREO COMPONENTS

LIKE YOUR COMPONENTS MOUNTED SEPARATELY? You'll find the very thing you want—from a tuner to a tweeter—in Stromberg-Carlson's catalog of high-fidelity components.

PREFER COMPONENT-QUALITY STEREO IN A HANDSOME ENSEMBLE? You'll find thirteen beautiful basic consoles in the catalog. You can have any one with the suggested high-fidelity components . . . or, in many models, with the combination of Stromberg-Carlson components that suits you best.

APPRECIATE VALUABLE, UP-TO-DATE, AUDIOPHILE INFORMATION? This colorful new catalog is loaded with facts and specs on the complete Stromberg-Carlson line—program sources, speakers, amplifiers, enclosures, turntables, speaker systems, ensembles—each of which is potent proof that "*there is nothing finer than a Stromberg-Carlson.*"

Get full data on such exciting new Stromberg-Carlson developments as . . .

PR-500 SINGLE-SPEED TURNTABLE

Earning raves from Hi-Fi publications for quiet action, constant speed and reasonable cost.

8-80 STEREO AMPLIFIER

Two 32-watt channels of continuous power. Compare it for quality with the two top names. Then look at the price tag.

Mail this coupon today for your free copy of the new Stromberg-Carlson full-line brochure.

Stromberg-Carlson Commercial Products Division
1476-09 North Goodman Street, Rochester 3, N. Y.

Please rush my free copy of your new line brochure.

Name ....

Address ....

City .... *Zone* .... *State* ....

I am particularly interested in ....

....

STROMBERG-CARLSON
A DIVISION OF **GENERAL DYNAMICS**

listener on the axis of symmetry.

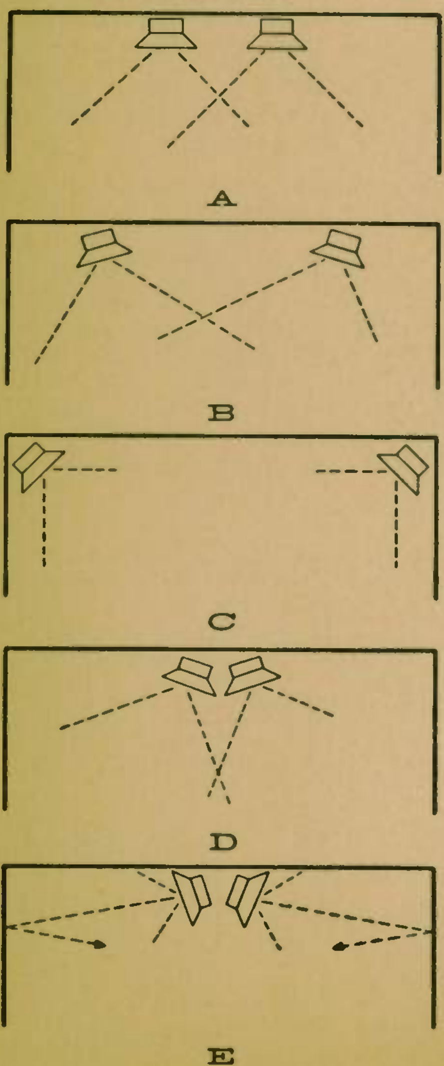
Phantom Center Speaker

The derivation of a phantom center channel has been the subject of a number of articles but seems to have been done first by Steinberg and Snow in 1933. The manner of derivation has been discussed in various publications.[2,3,4]

Qualitatively, the derived third channel resulted in retention of the geometric integrity of a string quartet and a large orchestra; a soloist standing some six feet to the left of the podium was reproduced a little to the left of the center speaker.

Quantitively, a study was made of the geometry of reproduction.[3] Sounds were generated in the pattern shown in Fig. 2A. With a two-speaker playback, an observer plotted the apparent sound sources as shown in Fig. 2B. With the three-channel playback using the derived center channel, a typical observation was that of Fig. 2C. As a "control" the same observer listening to the original sound (not over the loudspeakers) plotted the apparent sources as shown in Fig. 2D. The sounds were generated by a person speaking at the indicated locations, outdoors, and re-

Fig. 1. Arrangements used in experiments.



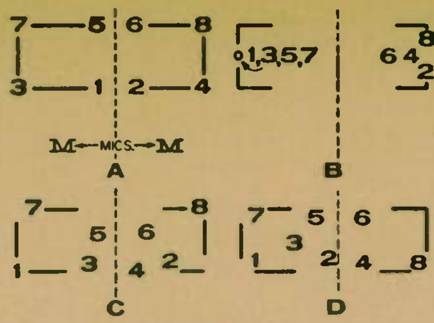


Fig. 2. (A) Actual locations of sound sources. (B) Apparent locations with two speakers. (C) Apparent locations with center speaker added. (D) Apparent locations during live listening experiment.

produced in a 16 foot by 25 foot room with speakers on the 25-foot wall. The results in Fig. 2D were obtained with the observer wearing a hood so he could see to plot his results but could not see the person speaking at the indicated locations.

Observers as much as 11 feet off-axis plotted results which were almost as accurate as those shown. This corroborates the work of Steinberg and Snow, indicating only small shifts of the virtual sources as the observer moves in front of a three-channel array.

Phantom-Channel Theory

The philosophy behind the phantom-channel technique may be stated as follows:

If two microphones are properly placed relative to each other and to the sound source, their combined output is that of a single microphone in the middle; this microphone "that wasn't there" can be reproduced by re-combination. The output of an actual third microphone can also be recovered by re-combination.

In practice, this combination may be accomplished by simple addition. The theory of the third channel derived from two sound tracks is still being developed, but it appears that crosstalk is subordinate to signal mutuality. (Crosstalk is the inadvertent transfer of signal from one channel to another; signal mutuality is the natural consequence of one microphone in a stereo array picking up signals pertinent to other microphones.[7])

The fact that the center channel carries sound from the flanks as well is true whether the channel is derived or independent. Experience shows that with proper adjustment of levels, a high degree of accuracy of geometric reproduction may be obtained with either the derived or independent center channel. As little as 2 db can produce a shift in the virtual sound source.

Speaker Placement

In the experiments involved with Fig. 2, the flanking speakers were placed in corners. This was deemed desirable for improved stereo geometry and also for improved tonality. Referring again to Figs. 1A, 1B, and 1C, no sound appears to come from outside the speaker array. Although the arrangement of Fig. 1E produced a "wide-stage" effect, it could not fulfill the requirements either of good geometry or good tonality. Thus, the arrangement of Fig. 1C plus a center channel was regarded as the only feasible array.

Monophonically the speaker response is improved by corner placement as a result of: 1. Complete room coverage with 90-degree tweeter radiation angle; 2. Better tonality or response; and 3. Accuracy in the lower three octaves of response.

Fig. 3 illustrates the benefits to be derived from corner placement—as far as tonality is concerned. A 15-inch driver unit in a 6.7-cubic-foot closed box on legs was tested four feet from the walls at a corner, on the floor in the same place, and on the floor in a corner. The curves show the responses for these various locations. Most no-

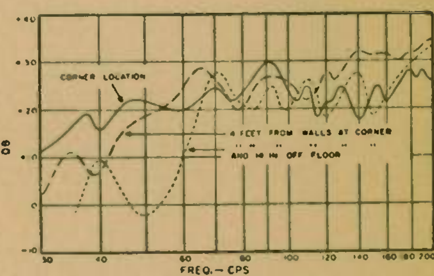


Fig. 3. Bass response of speaker system.

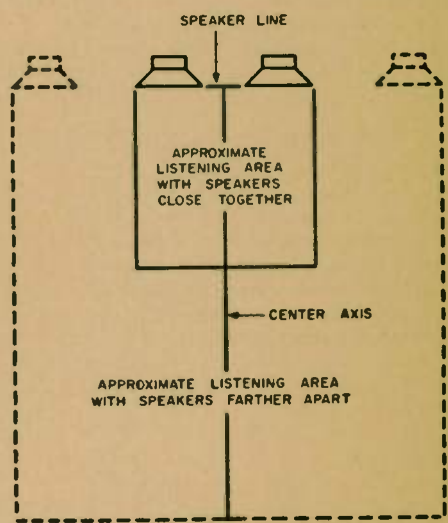
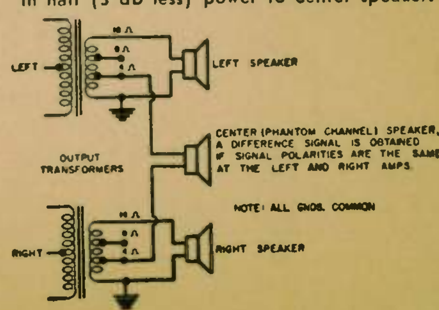


Fig. 4. Increase in area of stereo effect.

Fig. 5. Simple method of obtaining the phantom channel after the power amplifiers. Speakers have 16-ohms impedance and equal efficiency. Connection to 4-ohm tap results in half (3 db less) power to center speaker.



ticeable is the improvement below 60 cycles, but actually of comparable importance is the smoothing out of the 100-200 cycle range.

One must conclude from this that tonality and geometry demand corner placement of flanking speakers.

An experiment, not reported elsewhere, concerns a corner center speaker with wall-type flanking units, arranged in an "L" configuration. It was found that the geometry of reproduction was unnatural. Attempts to bring the center unit into proper geometry by increasing its signal input had the effect of causing it to "jump forward" into monophonic prominence. It is believed that the delay effects of some 5 to 10 milliseconds cannot be compensated successfully by increasing the volume, at least experience thus far negates the use of this configuration.

Corner placement permits the maximum separation and consequently the maximum listening area. The listening area is proportional to the square of the distance of speaker separation. (Refer to Fig. 4).

Outdoors, two speakers seven-feet apart could be detected as "stereo" at a distance of more than 50 feet.[5] Indoors, the distance decreased, with 14 feet providing *barely discernible* stereo effect. The maximum satisfactory listening distance was about 7 to 10 feet. Wider speaker placement insures adequate angle while addition of the center channel insures proper focusing so that the angular stage width becomes that of the original sound. Recall that the string quartet and soloist were properly located on playback as well as with a large orchestra.

Microphone Placement

Early stereo demonstrations appear to have concentrated on spectacular effects rather than reproduction of true stereo geometry. One such appears to have achieved a "three-peephole" playback effect by placing three microphones too close to the three separated sound sources.

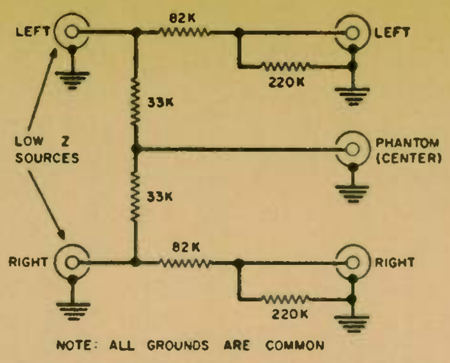


Fig. 6. Deriving the center channel.

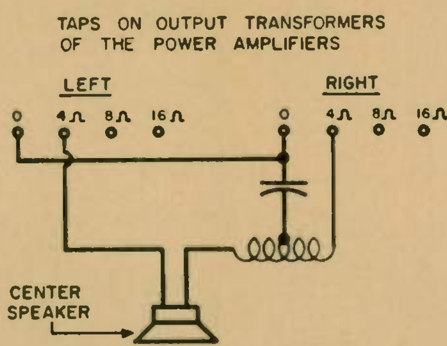


Fig. 7. Method proposed of combining the same polarity signals without cancellation.

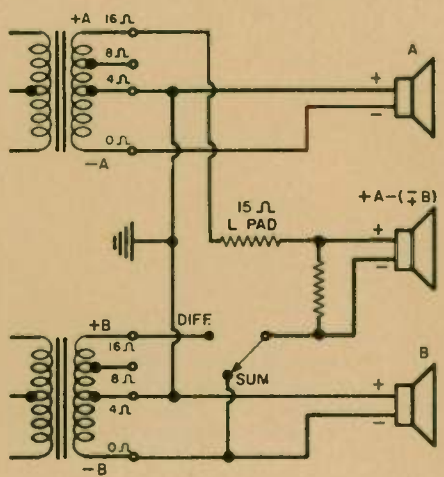


Fig. 8. This circuit may upset feedback ratio and cannot be used in all amplifiers.

Most current tapes and discs apparently have been cut using microphone placement which is compatible with three-channel playback. The bulk of this author's experience has been with two microphones. The current trend toward recording three sound tracks and later dubbing these to two involves a technology which is an art and science in itself. It is possible that microphone techniques which are capable of improving two-channel playback will offer even greater benefits in playback using three channels.

Deriving the Phantom Channel

The re-combination to derive a center channel may be accomplished in various ways. The original circuit[2] is shown in Fig. 6. This represents a "sum" combination while the "difference" circuit is shown in Fig. 5.

Systems using only two power amplifiers are based on intrinsic amplifier stability (precluding types using "damping control" or other forms of positive feedback).

Some recordings have been encountered in which focus of the center channel required a "sum" re-combination while others required a "difference" treatment. This option may be taken using the circuits of Figs. 7, 8, and 9. Fig. 9 employs the *Electro-Voice* XT-1 1:1 transformer. Fig. 7 involves the use of a special coil which is still in the experimental stage and not currently in production. It is believed the frequency at which 90-degree phase shift occurs should be placed at about 100 cycles.[6]

Fig. 8 derives the "sum" or "difference" without the exciting current and possible distortion of an additional transformer or coil. Fig. 10 shows a "sum" signal derivation using a preamplifier which permits a polarity reversal. ("Polarity" is used here rather than phase. "Phase" is the angular relation between two directed quantities where the angle may be any value while polarity applies to the special case where phase angles are confined to

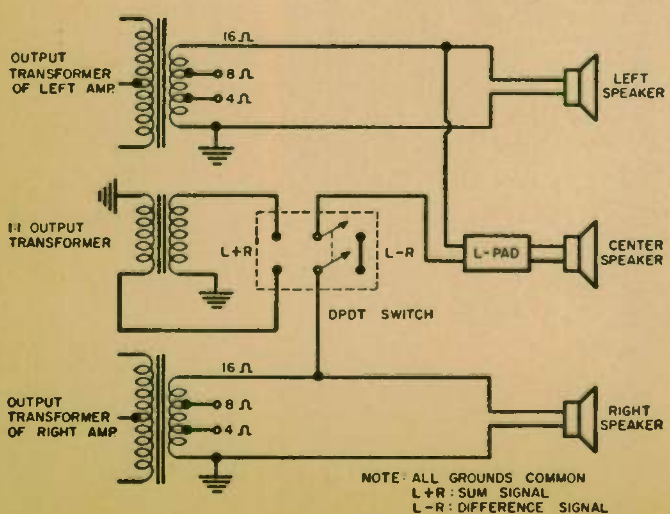
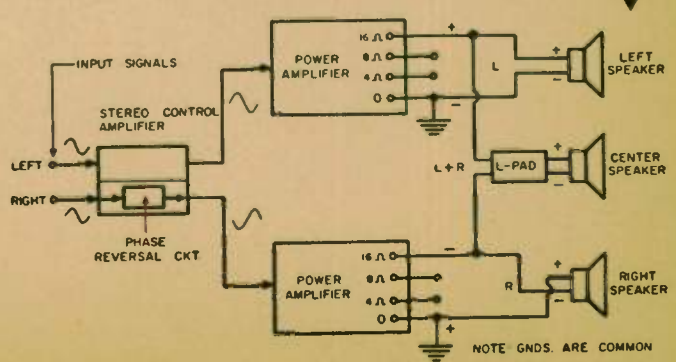


Fig. 9. Another method proposed by the author to obtain a sum and difference phantom-channel signal with 1:1 transformer.

Fig. 10. Method of obtaining a sum signal from two power amplifiers by first reversing phase of one of the input signals.



0, 180, and 360 degrees.)

All of the circuits, except that employing three amplifiers, assume speakers of approximately the same sound pressure output per volt of input. Impedance mismatches have been made in "tolerable" directions and assume speakers of 16-ohm nominal impedance. Output difference up to 6 db may be compensated by choice of output taps in two-amplifier systems. Speakers need not be of equal "efficiency" or output per *volt* input, but may differ as much as 6 db even in two-amplifier systems. Where a pad is indicated, the "L" pad is to be preferred over a "T."

The theoretical level of the center channel has been derived as 3 db down from the flanking channels[2], but experience shows this to be a function of environment. Room geometry has dictated center-channel levels from 0 to −9 db relative to the flanking channels and these values may not include all extremes.

Latest Experiments

Stereo geometry experiments have been conducted comparing three independent channels with two-track-derived three channels.[7] These experiments are still under way but those completed thus far indicate the two-microphone, two-track, three-channel system approaches the three-microphone, three-track, three-channel system in performance and exceeds the two-channel playback in accuracy of geometry.

This author is in agreement with Steinberg and Snow's[1] conclusions, *i.e.*, that the center channel is necessary for the preservation of a reasonable approximation of the original geometry in stereo playback. Addition of the center channel permits wider spacing of flanking speakers, culminating in the natural limiting case of corner placement and the natural angular rotation of flanking units for complete coverage of the wide listening area. Wide-stage stereo means wide listening area as well and the corner-limited arrays permit full advantage to be taken of the improved tonality afforded by corner-placed speakers and, preferably, corner-designed speakers.

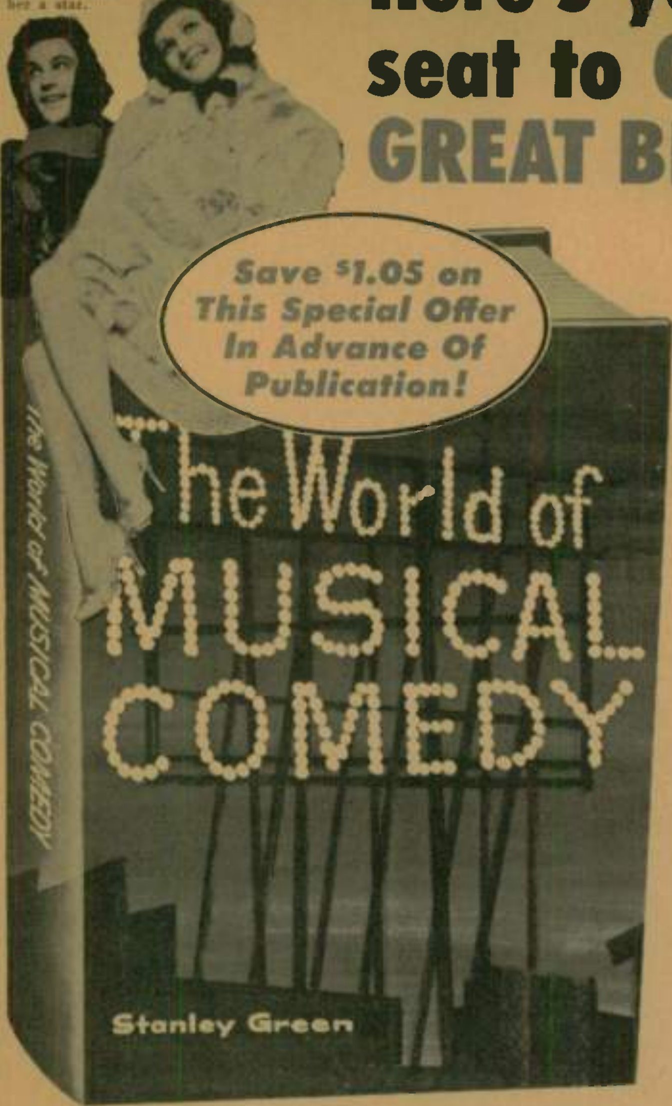
REFERENCES

1. Steinberg, J. C. & Snow, W. B.: "Symposium on Auditory Perspective—Physical Factors," Electrical Engineering, January 1934.
2. Klipsch, Paul W.: "Stereophonic Sound with Two Track, Three Channels by Means of Phantom Circuit, (2PH3)," Journal Audio Engineering Society, April 1958.
3. Klipsch, Paul W.: "Wide-Stage Stereo," IRE Transactions on Audio, July-August 1959.
4. Klipsch, Paul W.: "Circuits for Three Channel Stereophonic Playback Derived from Two Sound Tracks (10 ways to do it)," IRE Transactions on Audio, November-December 1959.
5. Klipsch, Paul W.: "Corner Speaker Placement," Journal Audio Engineering Society, July 1959.
6. Klipsch, Paul W.: "Three-Channel Stereo Playback of Two Tracks Derived from Three Microphones," IRE Transactions on Audio, March-April 1959.
7. Klipsch, Paul W. & Avedon, Robert C.: "Signal Mutuality and Cross Talk in Two- and Three-Track Three-Channel Stereo Systems," paper delivered before Audio Engineering Society Convention, October 8, 1959.

Here's Mary Martin (with Gene Kelly) about to sing *My Heart Belongs To Daddy*...in the show that made her a star.

Here's your front row seat to OVER 400 GREAT BROADWAY SHOWS!

An Extravaganza of Today's and Yesterday's Most Exciting Shows, Show People, and Show Tunes Thrillingly Captured in Story and Pictures



Save $1.05 on This Special Offer In Advance Of Publication!

GIANT LIBRARY SIZE – 374 PAGES – OVER 200 PHOTOGRAPHS – SPECIAL REFERENCE SECTION listing opening dates, number of performances, authors, composers, leading cast members, principal songs—plus complete record collector's buying guide of 12" long-playing records. An indispensable book for everyone connected with MUSIC and THE THEATRE.

NEVER before has there been a book which so vividly brings to life the magical sights and sounds ...the stars, dancers, and showgirls ...the wonderful singable tunes of America's musical theatre!

You'll re-live nostalgic favorites such as *Babes in Toyland*, *No, No, Nanette*, the *Ziegfeld Follies*, and other classics of bygone days. You'll thrill to *Pal Joey*, *Bloomer Girl*, *Finian's Rainbow*, *South Pacific*, *Carousel*...all the exciting, tuneful shows of the 30's and 40's. You'll cheer the spectacular hits of today—shows like *My Fair Lady*, *Music Man*, *Take Me Along*, *The Sound of Music*, *Fiorello!*...even "off-Broadway" musicals!

And that is not all. You also step backstage to get intriguing glimpses of the theatre at work—of life *behind* the footlights! You'll get new insight into the music of Victor Herbert, Sigmund Romberg, Irving Berlin, George Gershwin, Cole Porter and all the other "greats." You're there when composers and playwrights like Rodgers and Hammerstein, or Lerner and Loewe, write songs that enchant audiences the world over. You see stars like Judy Holliday, Rosalind Rusell, Pat Suzuki, Gwen Verdon create great roles out of their own, inimitable blend of art and personality...watch Ray Bolger, Marilyn Miller, Fred Astaire develop their exhilarating dance routines... get to know actors like William Gaxton, Alfred Drake, and Anthony Perkins...see Bert Lahr work out his funny "bits"...chuckle at Phil Silvers, Stubby Kaye, and Andy Griffith. Here's a galaxy of *all* the bright stars of Broadway—living portraits of hundreds of America's favorite performers!

This fabulous book will entertain you by the hour...with amusing incidents, personal highlights, rare photographs—and above all with the rousing parade of musical comedy from its very beginning right up to the present time. It's a delightful, heartwarming book...and one that you won't be able to put down without smiling – whistling – humming!

And if you're looking for *factual* information, it's all here: show titles, authors and collaborators... date of opening and length of run ...principal songs...leading cast members and their replacements... even a vast, informative listing of recordings that will help you choose the finest available records of musical shows for your collection!

SAVE 10 PERCENT ON SPECIAL PRE-PUBLICATION OFFER!

"The World of Musical Comedy" is now on press. After publication its price will be $10—but NOW, if you mail the coupon below, we'll send you a copy for a 7-Day Free Trial Examination. Read the book, and if you're not completely delighted, return it. You pay nothing, owe nothing. If you wish to keep the book, you'll be billed—not the regular retail price of $10.00—but the special pre-publication price of only $8.95 plus postage—a full saving to you of ten percent! Mail no-risk coupon below today!

You'll enjoy reading about the old-time shows and actors such as Paul McCullough and Bobby Clark.

MAIL THIS 7-DAY FREE TRIAL COUPON TODAY!

You'll find many intimate profiles of your favorite screen and TV personalities. Photo shows Walter Pidgeon and Jackie Gleason in TAKE ME ALONG.

You'll be surprised to know that...

George M. Cohan's rousing "You're a grand old flag" originally was "You're a grand old rag"!

A Cole Porter show containing a shipwreck had to be rewritten overnight because of the *Morro Castle* disaster!

Howard Dietz, the lyricist, created the M.G.M. lion—by borrowing Columbia University's emblem!

West Side Story was originally conceived as a musical called *East Side Story*!

Mary Martin was auditioned for Eliza Doolittle in *My Fair Lady*... but she didn't like the songs!

The estate of Vincent Youmans holds 175 unpublished and unused tunes by the composer!

One of the hit songs in *Gigi* was cut from *My Fair Lady*!

Ziff-Davis Publishing Company
One Park Avenue
New York 16, N. Y.

I want to take advantage of your pre-publication offer. Send me, on publication, a copy of THE WORLD OF MUSICAL COMEDY for 7-day free trial examination. I understand that if I am not completely delighted, I may return the book and owe nothing. Otherwise I will remit not the regular price of $10.00, but the *low pre-publication price of only $8.95* plus postage and handling.

1707

NAME________________________________
(please print)

ADDRESS________________________________

CITY________________ZONE______STATE__________

☐ SAVE! Enclose $8.95 now with this coupon, and we'll pay all shipping charges. Same return privilege and prompt refund guaranteed if not 100% delighted!

Tone-Arm Tracking Error in Stereo

By **FRANCIS A. GICCA**
Raytheon Co.

A serious loss of the stereo effect occurs with tracking error. Here is how this may be minimized by proper mounting of straight or offset tone arms.

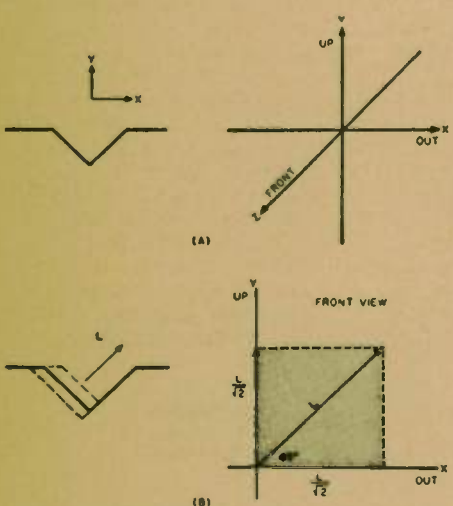


Fig. 1. (A) Groove coordinates. (B) Vector representation of left-channel output.

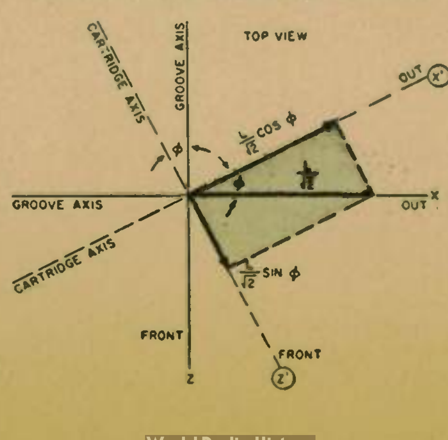
EDITOR'S NOTE: *This article deals with the horizontally measured tracking error. Equally important is the effect produced by vertical misalignment of the stylus. For information on this topic we refer our readers to the article "Pickup Cartridges for Stereo" by Norman Crowhurst in our October, 1958 issue.*

IT IS generally well known that unless a phonograph cartridge is at all times tangent to the groove of a monophonic recording, distortion is produced. With any pivoted pickup arm it is obvious that the angle between the axis of the cartridge and the tangent to the groove must change as the cartridge moves across the record. This angular error between the axis of the cartridge and the tangent to the groove is called "tracking error" and should always be minimized in order to reduce distortion. (For a discussion of tracking distortion see B. B. Bauer's article, "Tracking Angle in Phonograph Pickups," *Electronics*, March, 1955.)

However, it is not as generally known that tracking error also produces a serious loss of stereophonic effect when reproducing stereo discs. For example, if a 9-inch-long straight pickup arm is oriented so that the cartridge's stylus passes over the center of the disc, a loss in stereophonic effect of 31 db will occur in the outer grooves.

In order to investigate the cause of this degradation, and how it may be minimized, it is first necessary to establish a set of three-dimensional reference axes centered about the bottom of a stereophonic groove. Fig. 1A shows the coordinate system chosen. The X axis runs from the inside to the outside of the groove, the Y axis runs up and down the groove, and the Z axis runs front to back. These axes are mutually perpendicular.

Fig. 2. Translation of groove X output into the cartridge coordinates. See text.



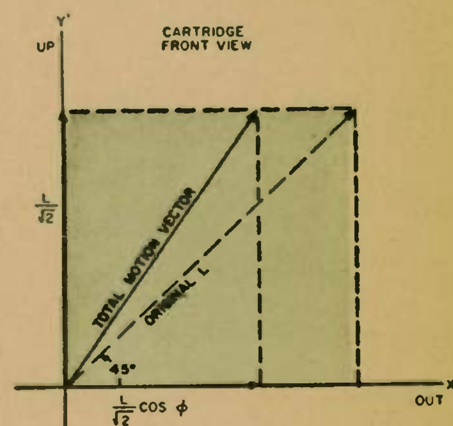


Fig. 3. Comparison between the total motion vector and the original L vector.

Assume that the disc is modulated with "left" output only, that is, the inner wall of the groove moves back and forth at a 45-degree angle to the horizontal as shown in Fig. 1B. This motion of the inner wall may be represented by a motion vector at a 45-degree angle in the in-out, up-down (X-Y) plane. Such a vector of length L has been drawn in the X-Y plane of Fig. 1B.

Since the groove wall is moving at a 45-degree angle, this is equivalent to equal lateral (in-out) and vertical (up-down) motions of the reproducing stylus. In other words, the L vector may be represented by two other vectors of equal length, one up-and-down and the other in-and-out. By trigonometry it is apparent that each of these vectors must have a length of $L \sin 45°$, or $L/\sqrt{2}$. These two vectors are shown in Fig. 1B.

If we look *down* upon the groove from above, we see the lateral in-and-out motion which the left modulation produces, but we will not notice the vertical up-and-down motion. In Fig. 2 we are looking down upon the groove (the X-Z plane) where we see the lateral $L/\sqrt{2}$ vector just found. If the cartridge's longitudinal axis is tangent to the groove, no tracking error exists and the cartridge's X-Z axes are the same as the groove's X-Z axes. In this case, the groove's lateral motion produces an equal lateral motion in the cartridge.

However, if a tracking error *does* exist, then the cartridge's X-Z axes will be rotated with respect to the groove's X-Z axes by a tracking error angle ϕ. The X'-Z' axes shown in Fig. 2 represent the axes of the cartridge rotated from the groove's X-Z axes, by the angle ϕ.

In order to determine the electrical output which the left groove motion will produce in a cartridge, it is necessary to determine the motion which the groove produces along the cartridge's output-sensitive axes. If there is no tracking error, then the left-lateral modulation produces only left-lateral motion in the cartridge. When tracking error is present, however, lateral modulation produces a front-to-back motion of the cartridge's stylus as well as lateral motion. The effect of this front-to-back motion is to reduce the effective lateral motion of the stylus. This reduction in lateral motion is a function of tracking error and can be found by projecting the lateral $L/\sqrt{2}$ modulation onto the cartridge's in-out (X') and front-back (Z') axes.

Fig. 2 shows this projection. By trigonometry it can be shown that the front-to-back (Z') motion is $L/\sqrt{2} \sin \phi$ and the in-and-out (X') lateral motion is $L/\sqrt{2} \cos \phi$. Since front-to-back motion of the stylus ideally produces no output, it may be neglected.

If we now look at a *front* view of the cartridge, we see the lateral $L/\sqrt{2} \cos \phi$ motion caused by the tracking error angle ϕ and the vertical $L/\sqrt{2}$ motion found previously in Fig. 1B. These two vectors are shown again in Fig. 3. If these two vectors are recombined into a single vector, it is seen that this total motion vector is no longer the same as the original L vector which produced it. This new vector is shorter than the original L and is *no longer at a 45-degree angle.*

Since the stylus motion is no longer at a 45-degree angle, the electrical output can no longer be exclusively left channel, but a crosstalk output is produced in the right channel as well. The result is a degradation in stereophonic effect since left output now appears in the right channel.

Mathematically, the left output is $L/2\ (1 + \cos \phi)$, and the right output is $L/2\ (1 - \cos \phi)$. If the tracking error is zero, that is $\phi = 0$, then $\cos \phi = 1$ and the left output becomes L and the right output is zero. If the tracking error is as severe as 60 degrees, then $\cos \phi = \frac{1}{2}$, which means the left output is $3L/4$, and the right output is $L/4$. This 3 to 1 ratio represents a separation between channels of only about 10 decibels. Of course, tracking errors of 60 degrees are generally not encountered, but it is possible to get tracking errors as high as 30 degrees with some currently advertised straight pickup arms mounted so that the stylus passes over the center of the disc. Because of the symmetry of the stereophonic groove, the situation is exactly the same for right-channel modulations.

A good figure of merit for the stereophonic leakage caused by tracking error is a *"crosstalk ratio" of left output to right output, expressed in decibels, when only left modulation is present on the disc.* Ideally, this ratio should be infinite which means that no right-channel output is produced by a left-channel signal. An infinite crosstalk ratio is achieved only when there is no tracking error. Since it is impossible to achieve zero tracking error across a stereophonic disc with a pivoted pickup arm, the arm should be mounted so that the crosstalk ratio is as high as possible across the entire disc. Fig. 4 illustrates how the crosstalk ratio varies as a function of track-

MANUFACTURER	OFFSET ANGLE (β, degrees)	ARM LENGTH (L, inches)	OVER-SHOOT ($-a$, inches)*	MAX. TRACKING ERROR (ϕ, degrees)	MIN. CROSSTALK RATIO (db)
Audax KT-16	19.50	12.19	.56	1.94	71.6
Electro-Sonic S1000	23.00	8.75	.60	2.27	68.2
Fairchild 282	23.00	8.94	.50	3.43	61.0
General Electric TM-2G	21.75	9.10	.54	2.27	68.2
Gray 212-SP	23.80	9.05	.68	1.75	72.6
Pickering 196 Unipoise	21.00	8.63	.50	3.65	58.8
Rek-O-Kut S-120 (12")	21.00	8.78	.53	3.63	60.0
Rek-O-Kut S-160 (16")	22.00	11.75	.75	2.02	70.2

* Note: The overshoot distance ($-a$) must be adjusted as accurately as possible in order to obtain these results.

Table 1. Tracking error, crosstalk for various stereo arms with typical cartridge.

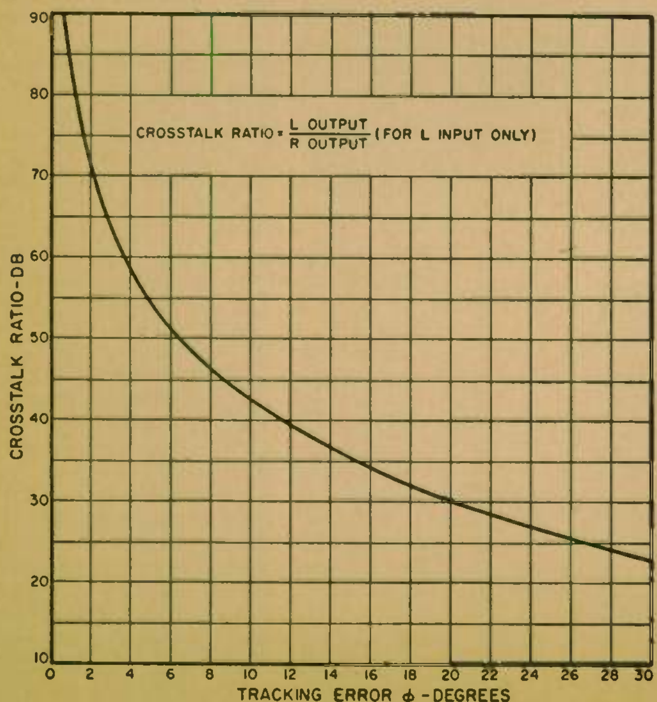
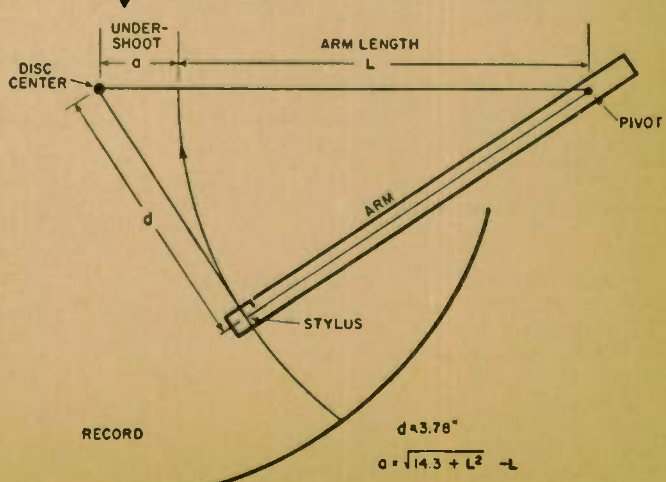


Fig. 4. Crosstalk ratio versus tracking error in degrees. Note that the crosstalk ratio is the relationship between the two output signals with an input to one channel only.

Fig. 5. The geometry of the straight tone-arm is illustrated in this diagram. Note that the arm must be mounted in such a manner that it undershoots the center of the record.



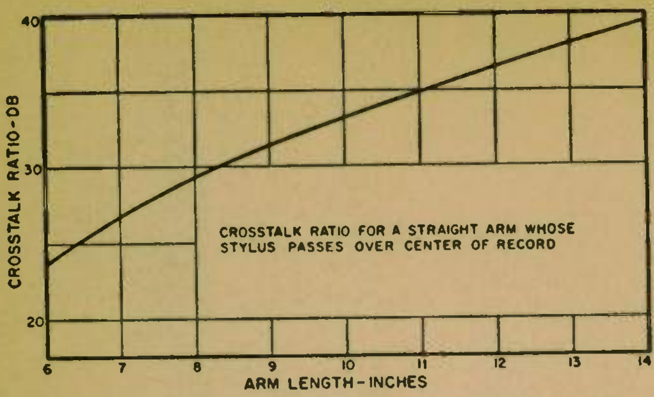


Fig. 6. Straight-arm crosstalk ratio versus length of arm.

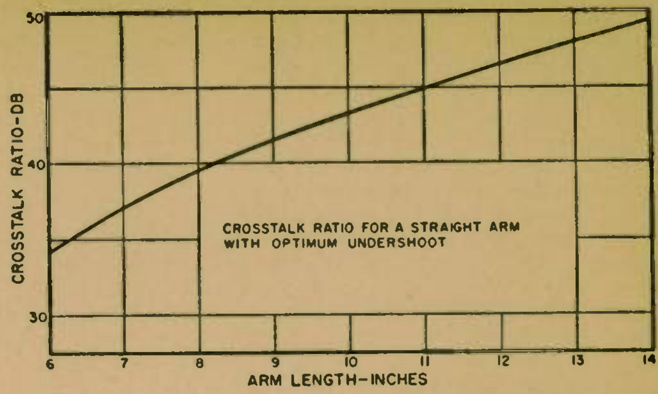


Fig. 7. Straight-arm crosstalk with optimum undershoot.

ing error from 0 to 30 degrees. Note that a tracking error of 30 degrees produces a crosstalk ratio of only 23 db, scarcely enough to maintain the stereophonic effect when the additional crosstalk already present in the disc, cartridge, and amplifier is considered.

In summary, tracking error causes a decrease in the lateral component in a stereo cartridge which, in turn, produces crosstalk between channels. The remainder of this article will analyze how tracking error may be minimized with straight and offset pickup arms so that the crosstalk ratio is maximized.

Straight Pickup Arms

Fig. 5 shows the general geometry for a straight pickup arm of length L. At some distance d from the center of the disc, the cartridge is tangent to the grooves. At this point the tracking error is zero, therefore the crosstalk ratio is infinite. As the arm moves away from this point, the tracking error increases and the crosstalk ratio decreases. In order to minimize tracking errors across the disc, the perpendicular distance d should be located somewhere in the middle of the disc so that the tracking errors are *equal* at the outside and inside of the disc. It is apparent that if d is located at the inside of the disc, the tracking error will increase tremendously at the outside. The same holds true for the inside if d is set at the outside. Minimum over-all error must occur if d is located somewhere between these two extremes, and it can be proven that minimum error occurs when the inner and outer tracking errors are made equal. Intuition also suggests this to be true.

Several 12" recordings were measured in order to determine at what maximum outer radius modulations started and at what minimum inner radius modulations ended. For a completely full, 12-inch disc, recording was started at a radius of 5.75 inches and ended at an inner radius of 2.50 inches. Setting the tracking errors equal at these two radii, yields $d = 3.78$ inches. Note that this value is the *geometric mean* of the two radii. This result means that all straight pickup arms, irrespective of length, should be tangent to the grooves at a distance of 3.78 inches from the center of the disc. Since it is difficult to determine tangency accurately, a more useful quantity is the undershoot a shown in Fig. 5, which specifies by how far the stylus should *undershoot* the center of the disc. The final result, see Fig. 5, is that a should equal $\sqrt{14.3 + L^2} - L$.

It is important to note that the stylus should *always* undershoot the center of the record for straight arms if optimum performance is to be achieved. Fig. 6 shows how the crosstalk ratio varies at the outer diameter of a stereo disc when a straight pickup arm is used whose stylus has been set to pass over the center of the disc, as a function of arm length. In contrast, Fig. 7 shows the crosstalk ratio that is achieved at the outer diameter if the correct undershoot is used. In both cases, the crosstalk ratio at the outer diameter is the worst condition. Note that optimum overshoot achieves a crosstalk ratio that is 11 db higher than the incorrectly positioned pickup.

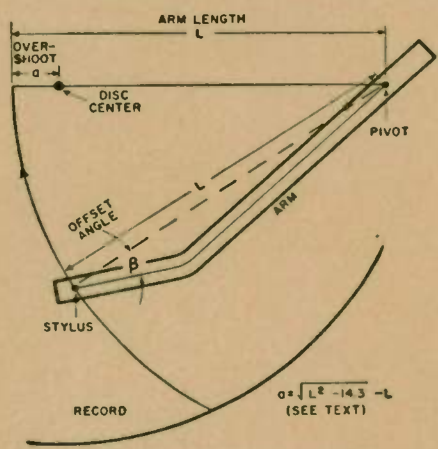
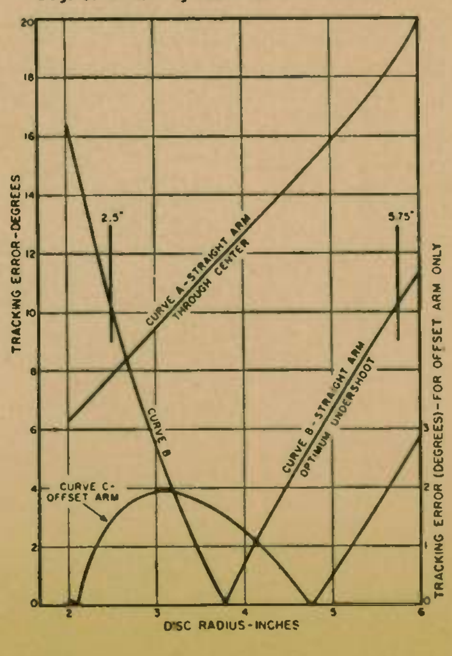


Fig. 8. Geometry of the offset tone-arm is illustrated in this diagram.

Fig. 9. Tracking error for two 9" arms.



Offset Pickup Arms

It is always possible to reduce the tracking error even further if an arm is used whose cartridge is offset from the arm axis by a small angle. Fig. 8 shows the general geometry for an offset pickup arm. As before, the length of the arm L is measured from the needle to the pivot point of the arm, which places L off the arm in the offset case. The offset angle β is measured as the angle which the lateral axis of the cartridge makes with L.

Using the same general analysis as for the straight pickup arm, a tangent distance d can be found for the offset arm which makes the tracking errors equal at inner and outer disc diameters, and this distance related to an optimum overshoot a. The result of this analysis for the offset case, however, is quite interesting and unique. It is found that the offset arm is tangent to the grooves at *two points* as the arm moves across the disc, rather than at just one point. It is also found that the tracking errors are equal at *three* points if the errors are set to be equal at the inner and outer disc radii as previously done. This third point turns out to be the *geometric mean* of the extreme radii, or 3.78 inches, as determined for the straight arm.

In other words, an optimum offset arm should be oriented so that it produces equal tracking errors at radii of 2.50, 3.78, and 5.75 inches. If we bear in mind the results of the straight arm analysis, it is clear that the offset arm should be designed and oriented so that it is tangent to the grooves at points which are the geometric means of the radii where equal tracking errors are desired. For equal errors at 2.50 and 3.78 inches, the arm should be tangent at 3.07 inches (the geometric mean). For equal errors at 3.78 and 5.75 inches, the arm should be tangent at 4.66 inches (again, the geometric mean). This analysis establishes the two tangency points for an offset arm to be at 3.07 and 4.66 inches. As before, an undershoot distance a can be found which will satisfy all of these conditions. The result is that a should equal $\sqrt{L^2 - 14.3} - L$.

There are two important things to note about this result. First, a will always be *negative* for an offset arm since the square-root term will always be smaller than $-L$. A negative value of a means that an offset arm always requires an *overshoot* rather than an undershoot. This is in contrast to the straight arm case which always requires an *undershoot*.

The second thing to note is that a *does not* depend upon the offset angle β. This is so because each length of offset arm automatically requires a specific offset angle which satisfies all conditions for minimum over-all tracking error. In other words, once an arm length is chosen by the manufacturer there is one, and *only* one, offset angle that will properly set the tracking errors equal at radii of 2.50, 3.78, and 5.75 inches. Any other choice of offset angle will set the tracking errors equal at completely different radii than those chosen in this article.

Unfortunately, there does not exist a general agreement among arm manufacturers as to what radii should be used in setting the tracking errors equal. The outer radius generally varies from 5.75 to 6.00 inches and the inner radius anywhere from 1.80 to 3.00 inches. These wide variations make it impossible to specify a simple formula which will allow the user to properly position any manufacturer's arm for minimum over-all tracking error. Therefore, each arm must be oriented so that it satisfies the tangency criteria which the manufacturer originally chose in the design of his arm. We can only question whether or not the manufacturer's choice is a realistic one.

Luckily, the offset arm is so superior to the straight arm that variations in design cause only minor variations in tracking error. To illustrate the superiority of the offset arm, Fig. 9 shows a comparative plot of the tracking error as two nine-inch arms proceed across a disc. Curve A shows the variation in tracking error for a nine-inch straight arm improperly oriented to pass over the center of the disc. Curve B is for the same straight arm re-oriented for an optimum undershoot of 0.77 inch. Curve C is for a nine-inch offset stereo arm (in this case, the *General Electric* TM-2G arm) mounted according to the manufacturer's instructions. Note the tremendous superiority of the offset arm. Perhaps the only criticism that can be made of this arm is that the tracking error is much larger at the outer radius than it actually need be. With a slight redesign, the outer tracking error could be reduced at a cost of inner tracking error. However, this would make the *over-all* tracking error much lower, which is to be desired.

In terms of crosstalk ratio, the incorrectly located straight arm has a crosstalk ratio of 31.6 db at an outer radius of 5.75 inches. When properly oriented, the crosstalk ratio increases to 42.1 db at this same radius, an improvement of 10.5 db. In sharp contrast, the offset arm has a crosstalk ratio of 68.2 db, a further increase of over 26 db!

Summary

The equation for optimum undershoot for straight pickup arms (Fig. 5) suggests that a straight stereo arm should be as long as practical in order to minimize tracking error and maximize stereophonic crosstalk ratio. However, a long arm must, of necessity, have a large dynamic mass which is undesirable for stereophonic reproduction since it causes excessive groove sidewall pressure. As a result, an offset arm is always preferable since this type allows tracking error to be minimized with a relatively short arm by proper design and selection of offset angle. In this case, then, it is desirable to have as *short* an arm as possible in order to lower the dynamic mass.

Fig. 8 includes the formula for optimum overshoot with an offset arm when the conditions of equal tracking errors at 2.50, 3.78, and 5.75 inch radii are present. Since most manufacturers *do not* use these conditions in the design of their arms, this equation is *not* universally useful but only helps indicate the general procedure used.

Table 1 gives the length, offset angle, and recommended overshoot for many commercially available stereo arms. The overshoot distance listed should be used for properly orienting the pickup arm. Using this optimum orientation, the resulting *maximum* tracking error has been calculated and tabulated along with the minimum crosstalk ratio at this point.

Note that in all cases the minimum crosstalk ratio is sufficiently high to guarantee a minimum of degradation of the stereophonic effect due to tracking error. This is the result of proper arm design *and* optimum orientation. Unless the arm is properly located for optimum overshoot, unnecessary degradation of the stereophonic effect will occur, and it must be remembered that this degradation is in addition to the crosstalk already present in the disc, cartridge, and amplifiers.

For arms not listed in Table 1, it is advisable to follow the manufacturer's recommendations carefully with regard to optimum overshoot or undershoot. It is wise to mount the pickup arm so that it may be moved readily when in place and the undershoot or overshoot distance accurately adjusted.

Stereo Microphone Techniques

By HERMAN BURSTEIN / **Information on mike placement to help make live tape recordings and to guide stereophile in speaker placement.**

AN ACQUAINTANCE with stereo microphone techniques can be useful to audiophiles in two ways. (1) It can assist the individual planning to make live recordings on tape. (2) It can guide the stereophile in the physical arrangement of his speakers inasmuch as certain microphone techniques are based upon a correspondence between microphone and speaker placement.

Stereo microphone techniques are based on general principles rather than hard and fast rules, because every recording situation has special characteristics and therefore a special solution. Nature of the sound source, acoustic characteristics of the performance site, and types of microphones employed are factors to be considered. Optimum results are thus obtained through a combination of experiment and adherence to general principles.

Binaural Recording

In the beginning, stereo was brought into the home on tape and was intended for reproduction *via* earphones. Therefore the early recording technique employed two microphones spaced about 8 inches apart and separated by a partition about the size and shape of the human head, as indicated in Fig. 1A. In effect, the microphones were an extension of the listener's ears. Since the microphone mounting method of Fig. 1A tends to be on the impractical side, arrangements such as that of Fig. 1B have been employed instead.

While binaural reproduction (*via* earphones) has given way commercially to stereo (*via* speakers), it is quite possible that some individuals

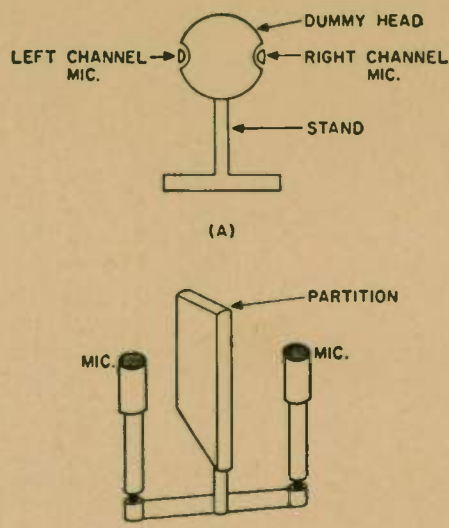


Fig. 1. (A) Dummy head for binaural recording. (B) A practical arrangement for such recording used in intensity-difference setup.

recording for their own pleasure and for reproduction through earphones—considered by many to provide a more realistic stereo illusion than speakers—will wish to use the microphone technique of Figs. 1A and 1B.

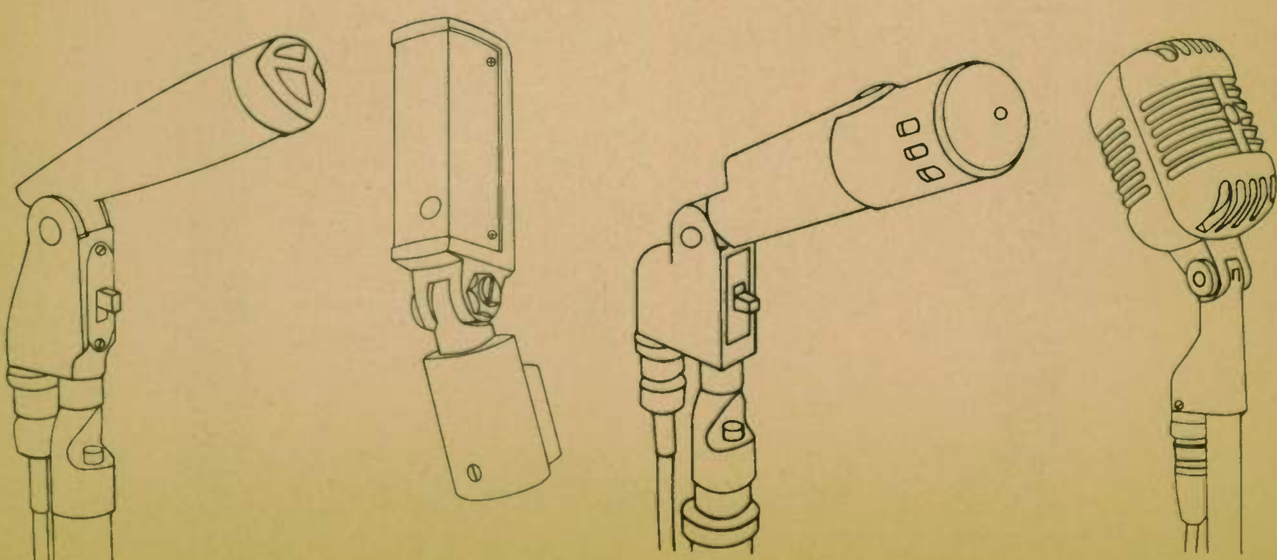
Time-Intensity Technique

When it became evident that the general public would not accept stereo through earphones but insisted on speakers, the microphones were spaced a matter of feet rather than inches apart. Spacing varied from about six to thirty feet. This is referred to as the "time-intensity" technique because it results in a difference in both the arrival time and the intensity of the sound at each microphone employed.

An early form of the time-intensity technique is referred to as "classical recording," where the microphones and the speakers were to be spaced about the same distance apart, with the distance depending substantially upon the breadth of the sound source, as represented in Fig. 2A. Actually, this is an application of the "curtain of sound" principle, where the sound source is spanned by a number of microphone-speaker links, each link covering a portion of the sound. In the case of Fig. 2A, the number of microphone-speaker links is limited to two.

The classical recording technique raises obvious difficulties. Because various sound sources differ in width, strict application of the technique would require not only the distance between microphones but also the distance between speakers to be constantly changing. The latter, of course, is impractical. However, the classical recording *principle* has been applied in a loose sense; microphones have been spaced in accordance with the nature of the sound source, without attempting to maintain the same spacing for the speakers.

Haphazard spacing of microphones along the source has given way to logical procedures which maintain left-right orientation and, at the same time, achieve a blend of the music as a whole. This can be achieved through the "listening-angle" principle, which involves a systematic relationship among microphones, speakers, and listener, as shown in Fig. 2B. The principle assumes an angle between approximately 30° and 45°, formed by a listener at the original performance



only for those who want the ultimate

SHERWOOD

"TOP RATED" again and again– and NOW AGAIN!

Model S-5000 20+20 watt "stereo"
Dual Amplifier-Preamplifier, Fair Trade Price—$189.50

Model S-2200 FM-AM-MX Stereo tuner,
Fair Trade Price—$179.50

AMERICAN AUDIO INSTITUTE AAI
394 EAST 18th ST. PATERSON 4, N. J.

October 27, 1959

Sherwood Electronic Labs., Inc.
4300 North California Avenue
Chicago 18, Illinois

Gentlemen:

We find that the incorporation of a center-channel output and a damping factor selector in July, 1959, increases the Summary Rating of the Sherwood S-5000 to the highest of all 18 Stereo Amplifiers tested in the AAI Evaluation Test Reports.

Sincerely,

AMERICAN AUDIO INSTITUTE

Felix R. Bremy

Felix R. Bremy
Executive Director

The "Most honored of them all" S-5000 stereo amplifier-preamplifier is joined by the S-2200 stereo tuner. As with its "Top Rated" predecessors, the S-2200 features FM "Interchannel Hush" plus push button selector, internal plug-in adaptor for Stereo FM Multiplex, 2 "Acro-beam" tuning indicators, simulcast FM/AM stereo. All Sherwood tuners feature FM sensitivity below 0.95 microvolts and ⅓% distortion @ 100% FM. For further details write: Sherwood Electronic Laboratories, Inc., 4300 N. California Avenue, Chicago 18, Illinois.

For complete specifications write Dept. AY-9 and the extremes of the sound source. The stereo microphones are mounted on the sides of this angle, and it is intended that in reproduction the listener and the two speakers shall form more or less the same angle.

The microphones can be placed at various points on the sides of the listening angle, so that their distance from the source and from each other will vary. Bringing them close to the source spreads them far apart, producing a marked left-right distinction, but creating the danger of a "hole-in-the-middle." Also, there will be a low ratio of reverberated-to-direct sound, with consequent loss of the effect of spaciousness. If the microphones are too far from the source, there will be little sense of directionality and the pickup of reverberated sound may be excessive, with consequent loss of clarity and definition. Between these extremes of microphone placement lies an area of satisfactory stereo results.

It might at first seem that the listening angle would be substantially different for a small sound source, such as a chamber group, than for a large one, such as an orchestra. However, a desirable listening position would ordinarily be much closer to a chamber group than to an orchestra, so that the angle between the listener and the source tends to be more or less the same in each case.

As a result of the distance between stereo microphones, there is a difference in arrival time of the sound at each microphone. In reproduction, according to expert opinion, this difference will produce the effect of directionality provided it is greater than 3 milliseconds, hence the microphones should be spaced at least three feet apart because sound travels approximately 1100 feet-per-second. On the other hand, it is possible for the difference in arrival time to be excessive. The ear tends to interpret differences above 50 milliseconds as two separate sounds rather than as a matter of directionality. Thus, if one is recording in a large hall, spacing of more than 50 feet between microphones is to be avoided.

When stereo microphones are placed far apart, there is apt to be a hole-in-the-middle, as previously noted, hence professional recording companies frequently employ a third microphone, whose output may either be blended with the outer microphones through mixers, as shown in Fig. 3A, or fed to the third channel of a three-channel tape recorder. In the latter case, the sound engineers can subsequently mix the center channel of the tape with the other two channels in the studio, which allows more time for achieving optimum results. Since three-channel tape machines are as yet far too costly for the average amateur, he must do his mixing at the recording stage.

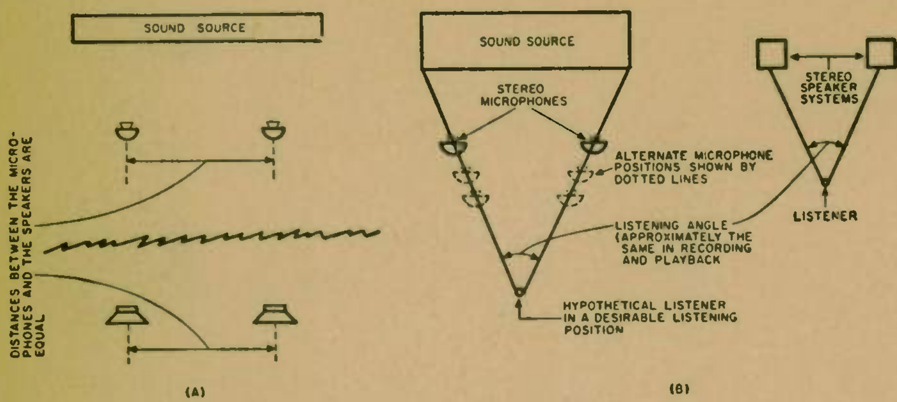


Fig. 2. (A) The classical recording setup. (B) The listening-angle principle.

Intensity Difference Technique

The "intensity difference" technique relies upon differences in signal amplitude at each microphone, but not on differences in arrival time. The microphone setup is essentially similar to that for binaural recording, with the microphones but a few inches apart, as was shown in Fig. 1B. Generally, microphones with a figure-8 polar characteristic are employed; these are most sensitive to sound arriving from the front and rear but not from the side. One microphone is pointed to the right side of the sound source and the other to the left side. The angle at which they are pointed is such that, between them, they also cover the center of the source. Since the microphones are closely spaced, the predominant difference in the signals arriving at each is one of intensity rather than arrival time. Use of a partition between them, as in Fig. 1B, increases the intensity difference.

A specific form of the intensity difference technique, called "Stereosonic" recording, utilizes two bi-directional microphones placed in the same vertical axis and arranged so that their polar characteristics are at a 90° angle to each other, as shown in Fig. 3B.

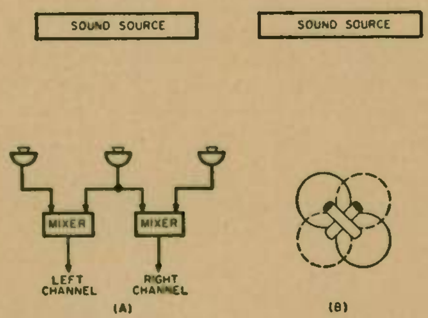


Fig. 3. (A) Use of a center microphone for stereo recording. (B) Stereosonic method.

Mid-Side Recording

A technique that has been widely used in Europe is known as "mid-side recording" and employs a specially designed microphone, the *Neumann* SM-2, which is known in the United States under the name of *Telefunken*. The SM-2 comprises two condenser microphones in a single housing. The "mid" microphone has a cardioid (heart-shaped) pickup pattern and is pointed at the sound source. The "side" microphone has a figure-8 pattern and is placed parallel to the source, so that the front receives sound from the left and the rear receives

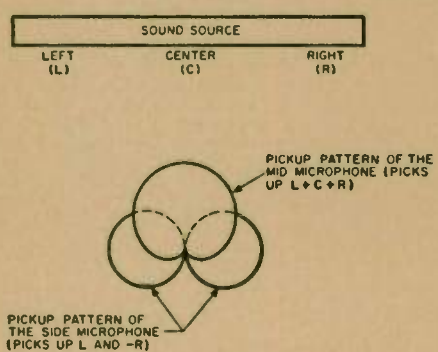
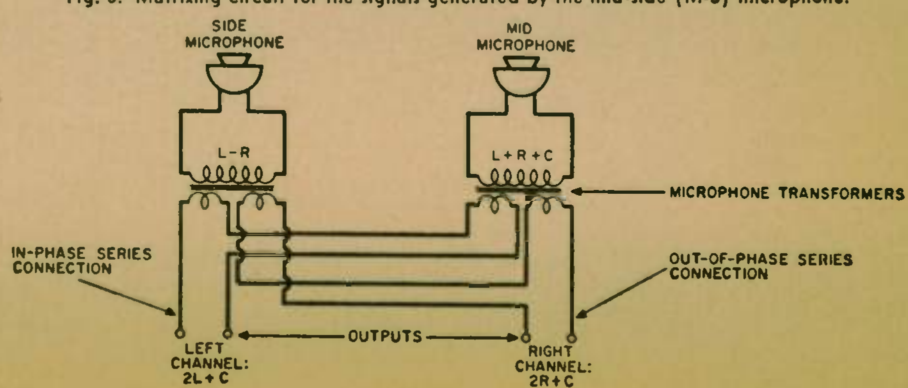


Fig. 4. Polar patterns of the two elements employed in the mid-side (M-S) microphone.

Fig. 5. Matrixing circuit for the signals generated by the mid-side (M-S) microphone.



sound from the right. The total pickup pattern with this technique is shown in Fig. 4.

The mid microphone picks up all the sound, namely from the left, right, and center, which we may designate as L+R+C. The side microphone has a single transducing element, which moves forward, say, when the sound is of positive polarity at the left and negative at the right. Hence the electrical signal produced by the side microphone is L–R rather than L+R.

The output transformers, which are part of the SM-2, have special windings that matrix the signals produced by the two microphones, as shown in Fig. 5. The L–R and L+R+C signals are combined in-phase to yield 2L+C. Also, the L–R and L+R+C signals are combined out-of-phase to yield 2R+C. In sum, one output contains primarily L information and the other output primarily R information, but between them they also reproduce the center of the sound source, thus eliminating the hole-in-the-middle effect.

While the SM-2 is priced beyond the budget of the average home recordist, there is the possibility that the mid-side technique may eventually be placed within his reach through a moderate-priced mixer that properly combines the signals of two low-cost microphones with the necessary polar characteristics.

Longitudinal Recording

In experimenting with stereo microphone placement, many things have been tried and one that deserves mention is the "longitudinal" technique, shown in Fig. 6A, where one microphone is placed behind the other. Two effects are achieved by this method. (1) There are differences in arrival time at each microphone. (2) There are differences in the ratio of direct-to-reverberated sound. Both effects create a sense of spaciousness. Moreover, in reproduction, each speaker delivers a different version of the original sound, and these two versions can blend to produce a fuller kind of sound than obtainable from either speaker alone.

It is possible to combine longitudinal recording with other arrangements. Thus one might employ two microphones at the left and right in usual stereo fashion, plus a third considerably farther back and in the center. The signal of the third microphone would then be mixed with the outputs of the left and right microphones.

Multiple Pairs

Sometimes more than one pair of stereo microphones is used, each pair having a different polar pattern in order to pick up sound in the desired manner. This is illustrated in Fig. 6B. Here a pair of cardioid microphones is used up front to pick up a soloist, while omnidirectional microphones are employed farther back to pick up the entire group of musicians and, at the same time, obtain an appreciable amount of reverberated sound.

Best results are apt to be obtained if the microphones are matched in a number of respects. Frequency response is one. The sound may tend to wander haphazardly between left and right in reproduction if the microphones used happen to have different sensitivities at the various frequencies being picked up.

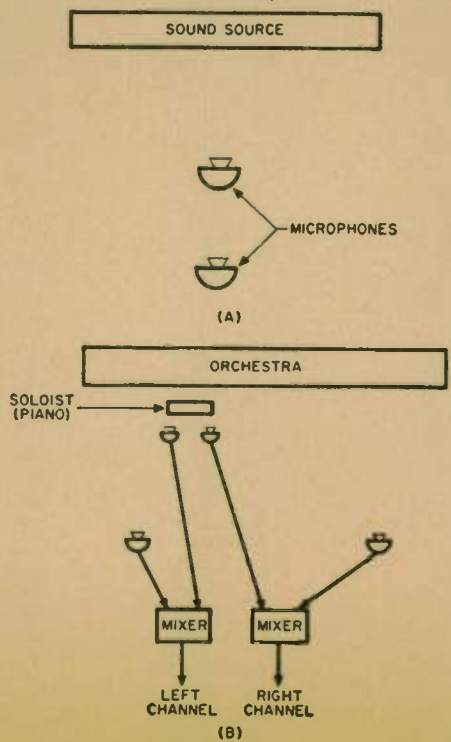
Matching of polar characteristics can be important. Apart from mid-side recording, the use of microphones with unlike coverage patterns may cause the apparent source of the sound to shift, or may result in inadequate reproduction of sounds in the center area.

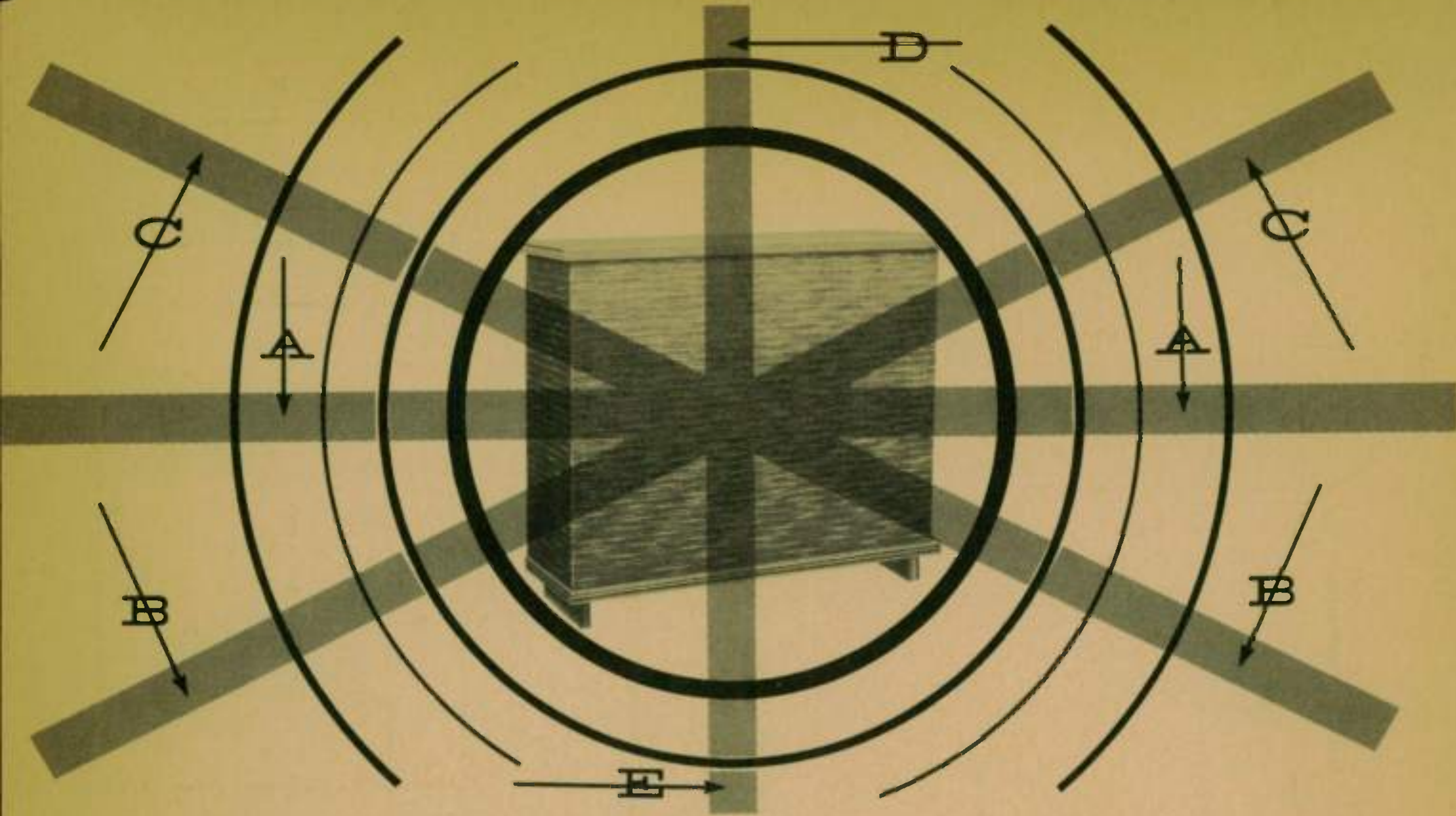
The microphones should also be matched with respect to over-all sensitivity if serious problems of channel balancing are to be avoided.

Phasing

The problem of correct phasing applies to the stereo microphones as much as to the rest of the stereo chain. There is good chance of incorrect phasing if one employs two different types of microphones. To establish whether the microphones are in-phase, one can do the following. Place the microphones next to each other, make a stereo tape recording of a steady sound (*e. g.*, from a test record), and play back the tape through a stereo system where proper phasing has already been established. If the sound appears to come from a point about midway between the two speakers, microphone phasing is correct. If the sound appears to come from an indefinite region, the microphones are not properly phased with respect to each other and it is necessary to reverse the leads of one of the microphones.

Fig. 6. (A) Longitudinal recording. (B) Use of two pairs of microphones for stereo.





Subdivided Hi-Fi Speaker Enclosure

By **D. F. REHBERGER**
Physics Dept., Univ. of Penna.

Complete construction details on unusual "infinite baffle" employing inexpensive speakers, based on a British design.

A GREAT deal of design and experimental manpower has been, and is being, expended in the improvement of low-frequency response in a given area—particularly with regards to cost. Considering size and price, most speaker-baffle combinations result in a compromise, with the low-frequency response below 500 cycles suffering. The major cause of poor low-frequency response, and one often overlooked in enclosure designs, is failure to remove low-frequency standing-wave effects from within the cabinet. Size and cost are limiting factors in conventional designs, as they usually go hand-in-hand so that the resulting system is either large or expensive, or both, without possessing the desired performance characteristics. Realistic reproduction is, of course, the prime concern but this can only be obtained after the basic problems of standing-wave effects and panel resonances are solved.

The purpose of the design to be discussed is to solve the major problems of enclosing a speaker in a baffle so that it produces good results, without the need for gimmicks or novelties. The design, basically a completely closed box or so-called infinite baffle, utilizes subdivisions as an acoustic filter, resulting in extremely good performance at a cost far less than the price normally paid for an enclosure to be used in the usual high-fidelity system. The size, too, has been reduced, being only 10-inches deep by 24-inches high.

The cabinet encloses a volume of air that is compressed and rarefied with the cone movement. This causes the bass-resonant frequency of the speaker to rise, due to the interaction of the speaker's resonant frequency with the resonant frequency of the air column. Besides the "air-column resonance," the reflective properties of the inside walls create standing waves at all but the high frequencies. Standing-wave effects with loudspeakers would be analogous to the standing-wave principle with transmission lines. Where there is no reflection from the load, in this case the interior of the cabinet, the sound waves would decrease exponentially toward the load. However, when the load reflects back part of the incident energy, it combines with the direct energy and varies, as to amplitude, periodically along the distance. This results in nodes (nulls) and antinodes (peaks). At a certain frequency and distance, with the rear of the cone located near an antinode, the output would be reduced by a large amount. The action of nodes and antinodes results in uneven response from the front of the cone and is attributed to standing-wave effects.

The reflecting sound waves (standing-wave effects) within the enclosure can be eliminated at all frequencies above about 500 cps by use of sound absorbing material within the cabinet. Below this frequency the standing waves cannot be eliminated successfully without some type of acoustic filter.

Panel resonances tend to color the program and produce noding if not constructed with extremely rigid panels. Good speaker enclosures of conventional design must be heavy and properly reinforced, with some being sand-filled or built with brick or reinforced concrete. Such a rigid cabinet would improve the transient response of the system, thereby resulting in a much cleaner output sound.

One of the early designs incorporating effective damping of standing waves was presented by D.E.L. Shorter of the *BBC* and consisted of subdividing partitions inside the cabinet (British Patent No. 696,671).[1] He states that the principle of subdivision may be achieved in various ways and the enclosure would have to be divided into many small sections, each of which would be unable to support standing waves at the low-frequency end. To prevent any reflecting-wave effects at higher frequencies, he suggests the use of sound absorbing material placed over two or more compartments around the back of the speaker. This prevents the cabinet from receiving any sound at high frequencies and eliminates the need for further acoustic treatment.

Of the many possibilities, the method used to subdivide the cabinet involved mounting radial partitions around the rear of the speaker. This not only accomplishes the necessary subdivision of the cabinet but also provides extremely rigid construction, re-

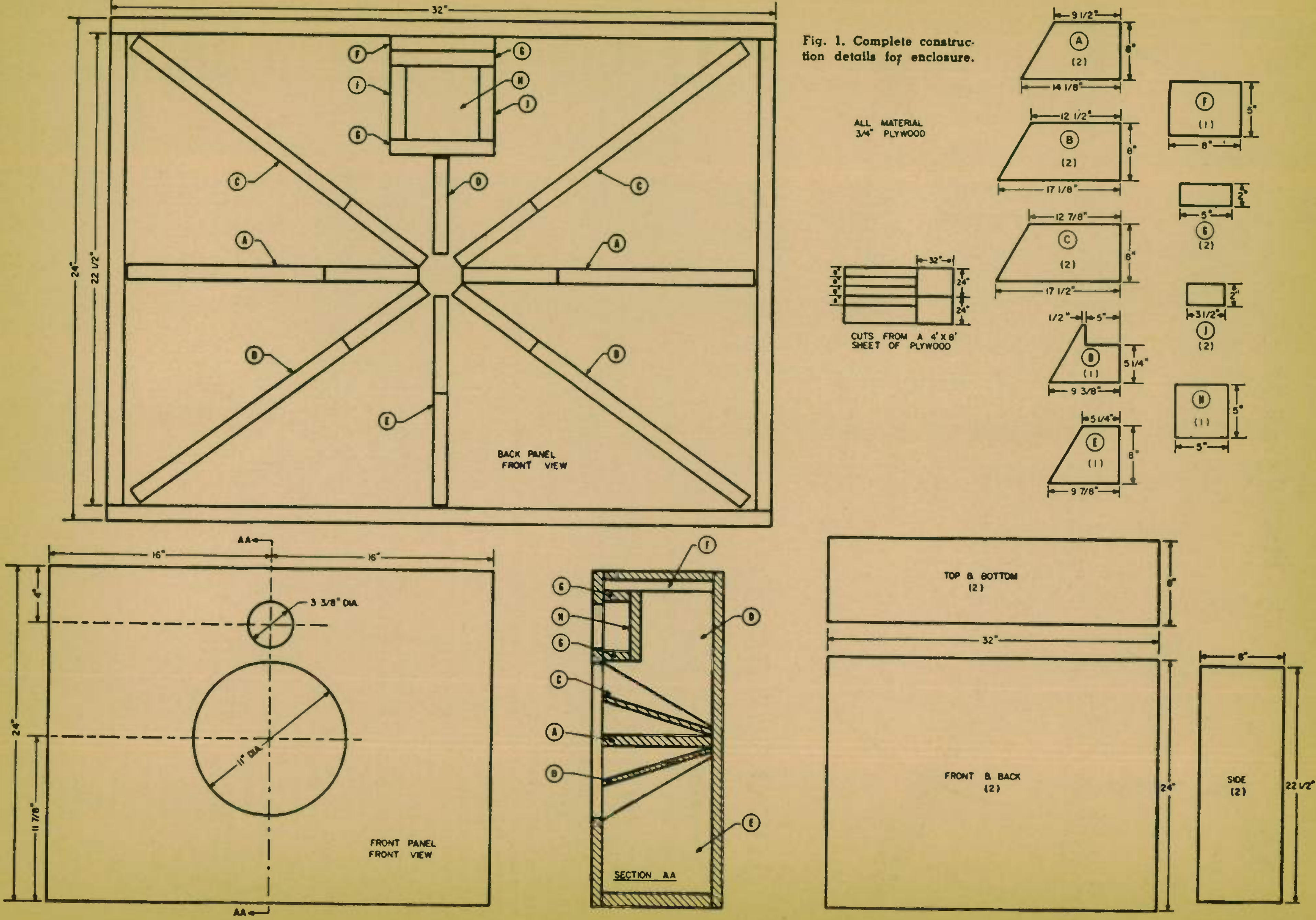
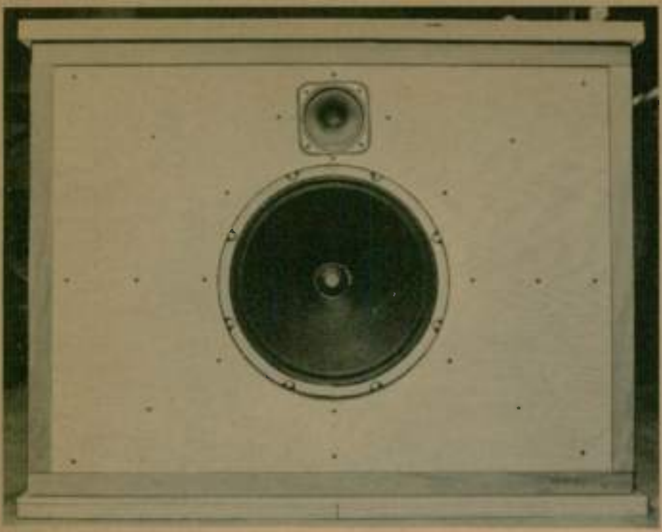
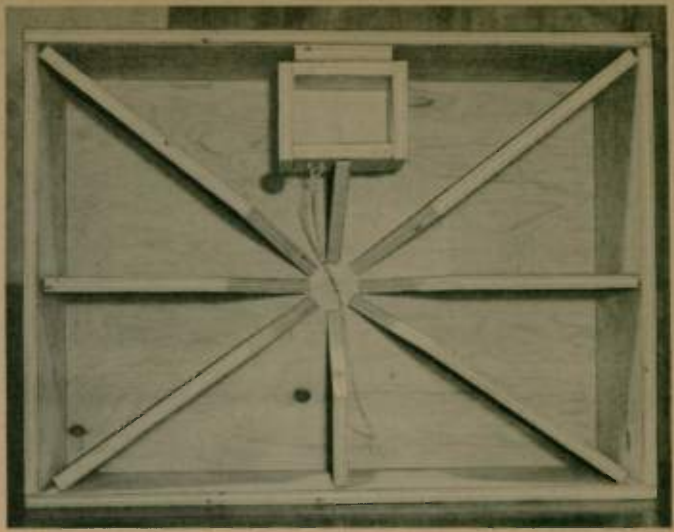


Fig. 1. Complete construction details for enclosure.



Front-panel view with speakers mounted, prior to being covered.



Front view of the back panel with the dividing partitions shown.

ducing the resonances of the panels.

The cone of the speaker used was treated, as suggested by Mr. Shorter, to obtain a reduction in the speaker's bass-resonant frequency by providing a freer suspension. The first model tested with the recommended speaker did not show any peaks below 1000 cps and it was felt that resonances were eliminated to a point where no further experimentation was necessary.

To obtain good, smooth response throughout the required range, without audible peaks or dips, the choice of speakers becomes fairly critical. Of the many 12-inch speakers tried, the *Jensen* P-12-P, which nets for about $17.00, was found to be most satisfactory with this enclosure. The tweeter is not as critical as the low-frequency unit but must have a smooth output without cone break-up and a level that matches the P-12-P. The *Jensen* P-35-VH, which costs about $4.00, matches the P-12-P and helps to provide the results obtained in the response curve of Fig. 2A. A simple high-pass filter is used, consisting of a 1-μf. capacitor in series with the tweeter, which makes up the "crossover network."

Treating the Woofer

A method of lowering the bass-resonant frequency can be applied to any large cone speaker that is not already treated (viscous edge damping) and provides a marked improvement. The cone edge, and only the edge, is impregnated with a hygroscopic material which will retain moisture in the cone edge and keep it more compliant. This results in a freer suspension and a lower bass-resonant frequency.

The cone edge is first treated with a wetting agent which provides a rapid path for the penetration of the hygroscopic material. The most convenient wetting agent is *Kodak* "Photo-Flo," available in four-ounce bottles at photographic supply houses. For use, dilute a bottle-cap-full of "Photo-Flo" in eight ounces of water (or 1-100 solution).

While the cone edge is still wet apply the hygroscopic solution. This is a saturated solution of calcium chloride (silica gel). The solution is made by dissolving as much calcium chloride as

The method of making the seal required for the woofer is illustrated above.

The placement of the sound-absorption material may be seen through woofer hole.

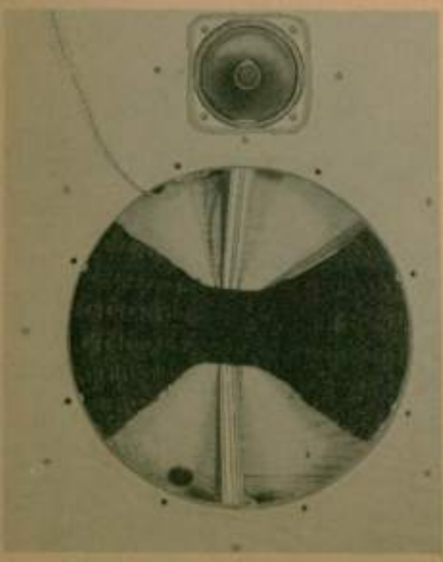
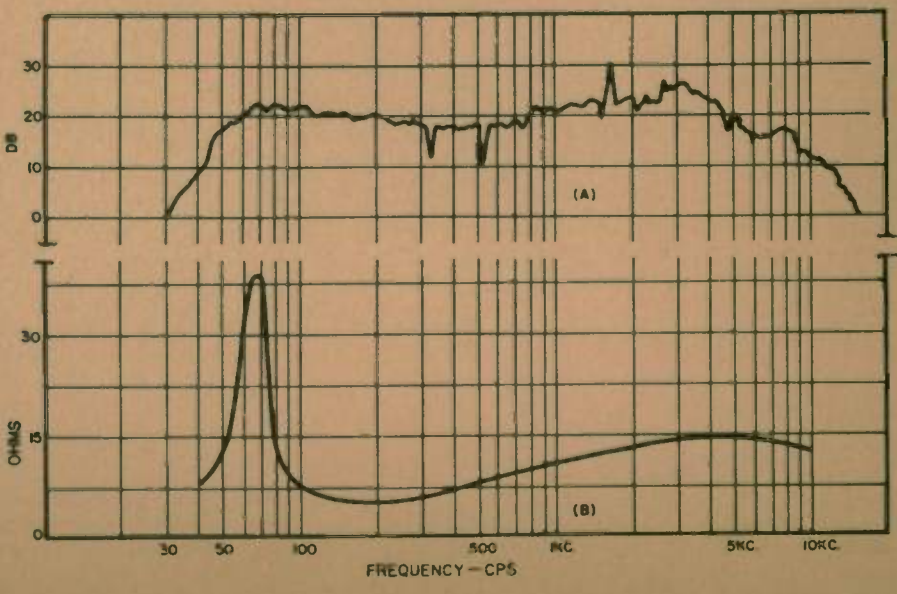


Fig. 2. (A) Acoustic output and (B) impedance curve of the speaker system described.



possible in about four ounces of water at room temperature. Allow the mixture to set a day and if none of the crystals has settled out of solution, dissolve more calcium chloride. A one-pound bottle of $CaCl_2 6H_2O$ should be enough, as only about half would be required. *Kodak* "Print Flattening Solution," used straight, would be an ideal substitute for the calcium chloride solution.

No matter which solution you pick, apply it with an acid brush or a paint brush. In the case of the calcium chloride solution, re-apply after about ten hours.

Because of the unique internal construction of the enclosure there is a need for a new method of speaker mounting and the final finishing of the complete enclosure. This can be accomplished by having the grille cloth removable instead of the back panel. All units built to date have been made of ¾-inch plywood, as this adds to the weight and rigidity. The enclosure can be made from a single 4 x 8 foot sheet and most lumber yards will make the cuts if the buyer doesn't have the equipment. The four pieces, 8 x 56 inches, must be within 1/16-inch on the 8" cuts, as these make up the top, bottom, sides, and partitions.

The entire enclosure is assembled using No. 5 flat-head woodscrews, 1½-inches long. White glue is used throughout to provide a good air-tight seal. The top, sides, and bottom are first assembled, followed by the back, partitions, and tweeter box. Wiring is completed next and then the front is secured in place.

The acoustic material is ordinary rug padding and is placed across two chambers on each side of the speaker as shown in the photograph. The 12-inch speaker is mounted in place with 10-24 "Tee-Nuts" (available at hardware stores) and round-head machine screws, one-inch long. If woodscrews are used they should be ¾-inch, No. 8 round-head. The tweeter is mounted with No. 6 round-head woodscrews, ¾-inch long. Both speakers are sealed with "Mortite," available at local hardware stores in four-ounce packages as shown in one of the photographs. Be careful, when tightening the screws, not to damage the cone or warp the chassis.

The most practical method of finishing the enclosure is to use grille cloth to cover both sides as well as the front. This reduces the total surface of finished wood to a minimum. The cloth is stretched around the cabinet and secured in place with wood strips and wire nails.

Testing the Enclosure

It is quite obvious that the final test of a speaker system is by objective listening and this is something that cannot be put into words, due to the many variables. This leads to a need for visual information, hence, the frequency response and impedance curves shown in Fig. 2.

The frequency response curves were taken in an average living room, using the procedure described by G. A. Briggs[2] and devised by Shorter.[1] The microphone was placed on the axis of the 12-inch unit at a distance of one foot and the system then covered with sound absorbing material to reduce room effects. Heavy wool blankets were used as the damping material and room effects were reduced sufficiently to provide a good compromise between an anechoic chamber and a live listening room. Response curves taken by Shorter in the arrangement described[2] were within 2 db below 1000 cps and within 1 db above 1000 cps of curves taken outside in free-field conditions. For response curves above 1000 cps, the microphone was moved back to three feet so as to include the output of both speakers.

The curves were plotted just as they appeared in the form of voltage changes converted to decibels, and peaks and dips were not altered in any way. This gives a representation of the acoustic output from 40 to 15,000 cps. Since the ear cannot detect narrow peaks or dips these can be ignored for all practical purposes, giving a much smoother over-all curve.

The two speakers are of different impedance, the woofer being 8 ohms and the tweeter 16 ohms. This difference helps give a smoother impedance curve (Fig. 2B), resulting in a better frequency response curve.

The transient response is very good, due to the absence of panel resonance and the use of an amplifier with a high damping factor. In order to realize the maximum performance of the system, it should be used with an amplifier having a damping factor of 5 or more. Most amplifiers using triodes or beam power tubes with feedback would be acceptable.

Since the methods of determining the efficiency of a speaker system are somewhat ambiguous, little will be mentioned here. This system is more efficient than most infinite baffle and reflex systems. It is safe to say that a 10-watt amplifier would be adequate for home use as only 50 to 80 milliwatts are required to provide a level of about 80 db (0 db = .002 dyne/cm²).

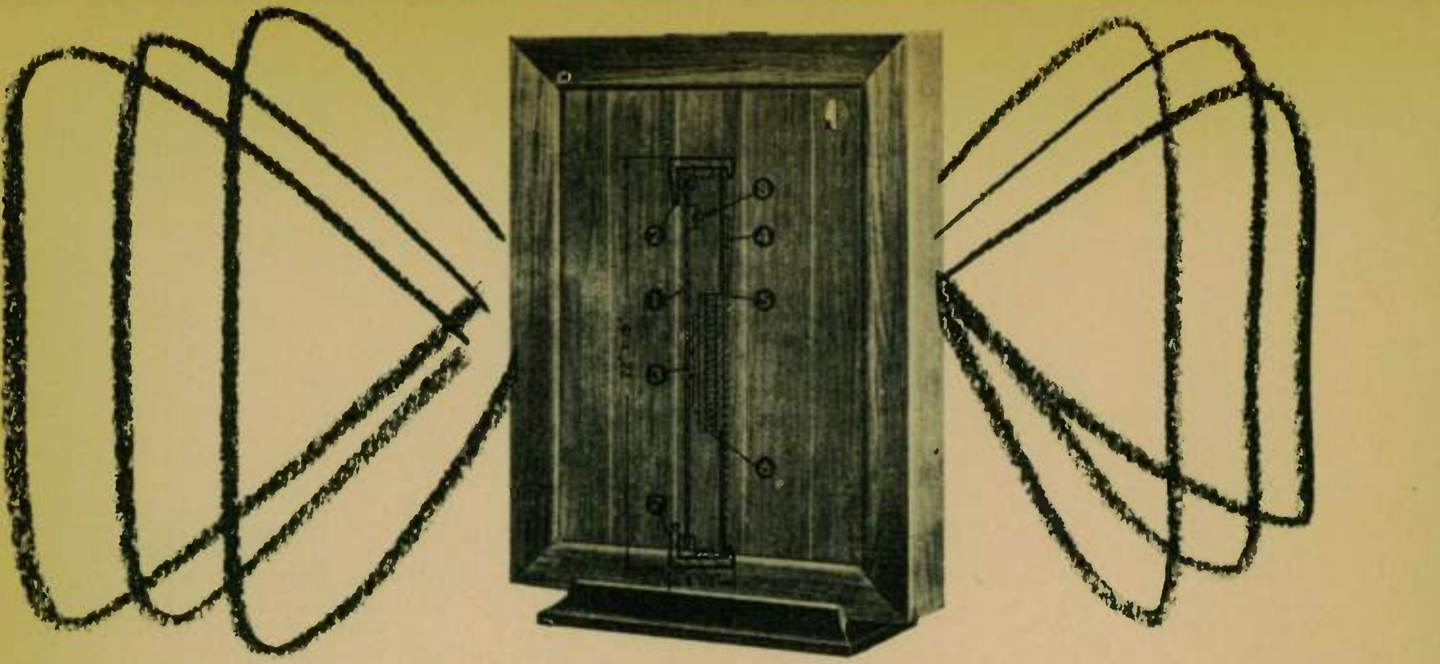
The results obtained certainly qualify this speaker system as a "high-fidelity" unit. Considering the initial cost, it is an excellent basic unit and two of them will make a fine stereo speaker setup.

In keeping with the wishes of the *British Broadcasting Corporation*, which has given the author permission to make use of its patent specifications, we wish to state that they are in no way connected with this specific design.

REFERENCES

1. *Shorter, D. E. L.: "Sidelights on Loudspeaker Cabinet Design," Wireless World, November 1950.*

2. *Briggs, G. A.: "All About Audio and Hi-Fi—Testing Loudspeakers" (Part 4),* RADIO & TV NEWS, *August 1957.*



The "Bi-Phonic Coupler"

A Unique Hi-Fi Speaker System

By **ABRAHAM B. COHEN**/President, Advanced Acoustics Corp.

An unbaffled system, using a large flat wooden diaphragm rather than a small paper cone to radiate hi-fi sound, promises fewer decor problems.

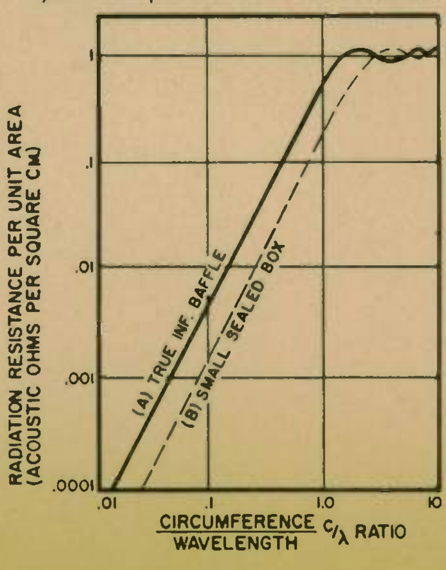
LOUDSPEAKER system design has undergone a period of evolution engendered, in the main, by the progress that has been made in the design of the "heart" of the system itself—the driver unit. In the earliest days of sound reproduction the driver unit consisted of an over-powerful earphone coupled to a "morning-glory" type horn. In fact, such a system was originally called a "loudspeaking telephone," which in due time became fore-shortened to simply "loudspeaker." The morning-glory horn of the early days was not a thing of living-room beauty, but this was not a matter of great concern, for its lack of decorative merit was compensated by the fact that it made possible the technical marvel of mechanically reproduced sound.

From these early beginnings, the loudspeaker mechanism has been improved to the point where any present advance is measurable only in the "last decimal place." This is not generally true, however, for the enclosures that up to now have been as important to the proper reproduction of sound as the loudspeaker itself. In the history of enclosure design we have gone through phases involving the large flat open baffle; the large closed box approximating the infinite baffle; the horn-loaded baffle; the bass-reflex baffle; the combined rear- and front-loaded enclosure; and, most recently, the small sealed box. These systems have ranged from large, massive, wardrobe-type of cabinetry that were awe-inspiring in size (and difficult of placement), to the very small enclosures that today are lumped into "bookshelf" category or system.

The one common characteristic of all of these systems is that, in some form or another, they are boxes that house a separate sound producer—the loudspeaker itself. These boxes must somehow be fitted into the general pattern of furniture arrangement and, because they are boxes, all have a "live" side, a "dead" back side, and must be used in such a fashion as to show their faces and hide their backs.

The loudspeaker system to be described—called the "Bi-Phonic Coupler," Fig. 2—in addition to being a radical departure from conventional design has broken through the "decor barrier." It is a *live* system in that it is not boxed. It radiates from both sides of its thin contour so that it may be placed, if desired, in the center of a room to fill the room with sound produced uniformly from both sides. Because it is thus freed from the boxed container, it is also freed from the conventional loudspeaker location areas, although it may, of course, be used in these conventional spots if desired.

Fig. 1. Radiation resistance seen by piston in small sealed box is about half that seen by the same piston in a true infinite baffle.



Of more interest to the technical man is the way in which the very deepest of bass and breadth of sound can be obtained from a system which is so radically different from conventional loudspeakers—a system which uses no conventional cone, is completely unbaffled, and produces its sound from the vibrations of a very stiffly held wooden panel which is, literally, as "stiff as a board."

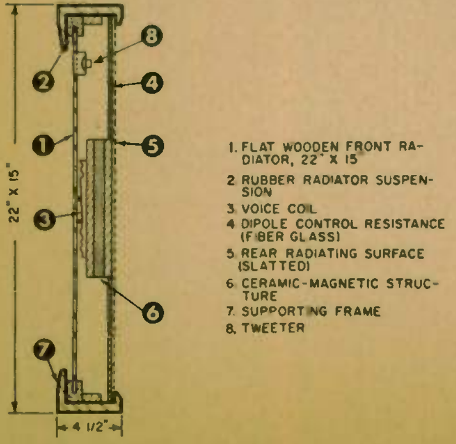
Loudspeakers generally reproduce sound by the vibrations of rigid "pistons," which are loosely suspended. In general, the cone-type loudspeaker makes use of a deep-formed molded, or folded, paper cone so that it may obtain, by this construction, some measure of rigidity while it is vibrating. Such a flexibly supported rigid cone structure vibrates and produces sound when driven by an electrical signal applied to its voice coil which is immersed in the magnetic circuit. The degree to which such a rigid piston produces sound is determined by its size and how it is baffled—a baffle signifying a device which prevents the rear wave of sound from the diaphragm from interfering with the front wave. Such a baffle, called an infinite baffle, is one which theoretically should be uniplanar with the piston surface and extends infinitely far in all directions of the plane. In actual practice, a large wall between two rooms with the piston mounted in the wall fairly well towards the *central* part of the wall very nearly constitutes an infinite baffle for there exists complete front-to-rear isolation of the sound vibrating from both sides of the speaker. And yet, because of the large volume of the rooms involved, the loudspeaker is radiating in relatively free space on either side.

Radiation Resistance

When a vibrating piston is mounted in such an infinite baffle, its sound output, at a given frequency, is governed by the radiation resistance it "sees" at that frequency. This dependence of the radiation resistance upon piston size and applied frequency is shown graphically in Curve A of Fig. 1. It will be observed that for a given frequency (or wavelength) the radiation resistance will vary directly as the radius of the piston. Since the acoustic power that the piston can deliver is a direct function of the radiation resistance it sees, then to produce an increase in radiated power at a given frequency (for a given electrical input power) the piston radius would have to be increased. This would increase the C/λ ratio and move the radiation resistance operating point higher up on the curve. This, basically, is why a 15" speaker produces better low-frequency radiation than a 12", for example, all other controlling elements remaining unchanged.

Thus, *theoretically*, it is possible to increase the radiation resistance for a given frequency at will by going to larger and larger diaphragms with increasingly better reproduction at low frequencies. Practically, however, there are physical limitations imposed on making conventional diaphragms too large. Molded paper cones become physically unstable and do not act as true pistons when they get too large—for they begin to flex within themselves. Second, the larger they get, the deeper they must be made to maintain some measure of the mechanical stability.

Fig. 2. Cross-section of speaker system.



Furthermore, with its accompanying magnet structure, the large size cone-type speaker becomes excessively large and deep necessitating an even larger and deeper box-type enclosure in which to house the speaker whose function it will be to baffle the rear-radiated wave from the front.

Curves are given in the literature which indicate that for a true infinite baffle, the radiation resistance, as seen by the piston mounted in that baffle, is approximately twice the value of the radiation resistance for the same in a small sealed box; usually smaller than 8 cubic feet (Curve B, Fig. 1). The reason for this is that although the small sealed box may prevent front-to-rear wave cancellation, the piston does not "see" the large, flat, infinitely extended baffle plane. Instead, the radiation diffracts around the small box edges and thereby loses some of its forward projection radiation. The small box radiates into "full acoustic space" of 4π steradians while the true infinite baffle radiates into "half space" of 2π steradians, with, of course, an attendant power gain when it thus radiates into only half the volume of the small box. Thus, simply sealing the back end of a piston in a box does not necessarily result in optimum acoustic coupling to space.

Now, let us consider the situation which would arise were we to free ourselves from the baffle entirely. Again, we look to the established literature and find that there is a curve of radiation resistance of a piston radiating without aid of a baffle at all. This curve (Fig. 3) merely verifies what we have learned from everyday practice, namely, that for a *given* size piston radiating at a *given* frequency, radiation efficiency will be greater when baffled than when unbaffled. Thus, for instance, if the piston is sealed off and operating at some frequency such that the ratio of piston circumference to wavelength is 0.35 (as in Fig. 3A), then the radiation resistance will be 0.015 acoustic ohms per cm². However, for the same piston unbaffled, operating at the same frequency, the radiation resistance drops to .0001. This effect simply leads to the well-known conclusion that when unbaffled the piston will not produce as much low-frequency power as when baffled.

Coupler may be placed vertically, horizontally, or even hung directly on wall.

Unbaffled Piston

However, there is another way of looking at this relationship whereby it will be possible to realize as much power from an unbaffled piston as from a baffled one. Using the same relationships, we are going *to choose a radiation efficiency level that we might expect from a boxed piston operating at a given frequency and find what size unbaffled piston will correspond to the same radiation resistance level.* Thus, for a given level of radiation, we simply move horizontally from the baffled curve to the unbaffled curve and can now find a different circumference-to-wavelength ratio for the unbaffled piston which will produce the same radiation output as the small baffled piston ratio of piston circumference to the same wavelength. From Fig. 3B, it will be seen that for a radiation resistance of .0015 for a C/λ of .1 (small box condition), we move horizontally to the unbaffled curve and find a value close to .5. Since, however, we are maintaining the wavelength, λ, constant, then the circumference of the *unbaffled* piston will have to be approximately five times as large as the baffled one to produce the same power output at that given frequency. Thus, by simply choosing the right size radiator, we may *theoretically* reproduce, by means of an unbaffled piston, any desired low-frequency power equivalent to that of a considerably smaller *boxed* piston.

However, for purposes of adaptability to home use, a piston size of 15" x 22" was chosen and even this size is not too critical. For example, if we use the figure of .0015 radiation resistance level for a circumference-to-wavelength ratio of .1 for a baffled piston and move horizontally to some intermediate value of C/λ such as .4, which for a fixed frequency means a piston four times as large as the baffled one, then we find that we are *almost, but not directly* on the curve representing an unbaffled piston. At this point of C/λ equal to .4, which represents an *almost unbaffled piston* condition, we may expect the same level of radiation as a baffled piston of one quarter its size.

Now the expression "almost unbaffled" may seem paradoxical. This is actually only a matter of semantics for what we mean by a closed box is really a device that prevents the rear wave of the speaker from coming around to the front. Actually, then, the phrase "somewhere between a closed box and an unbaffled piston" may be restated as "at some condition of radiation where *part* of the rear wave is kept from interfering with the front wave." With this in-

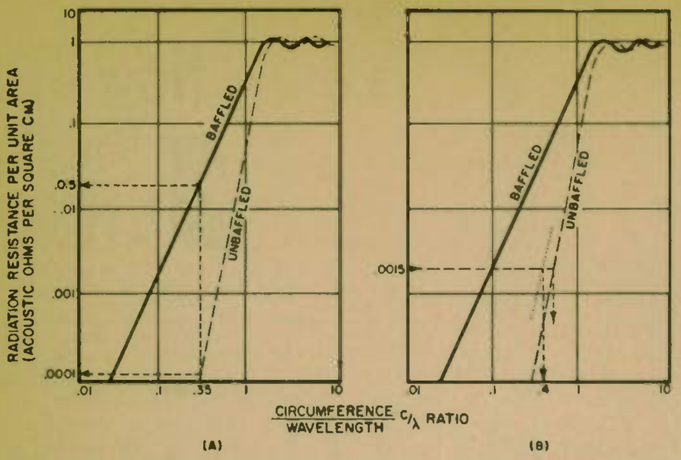


Fig. 3. (A) For a given wavelength, a fixed size piston will produce much less low-frequency power when unbaffled than when baffled. (B) However, for a given radiation level and for a given wavelength, a large unbaffled piston size may be obtained to produce the same low-frequency power as a smaller piston when in a closed box.

terpretation, it becomes fairly easy to provide the necessary control of the radiation from the back side so that the piston will operate on the intermediary characteristic indicated on the graph.

The rear-wave control is actually a combination of acoustic resistance material and an aperture screen which, together, provide just the right attenuation of the rear radiation to allow the free piston to produce the necessary output power without aid of conventional baffling. By actual measurement the rear attenuation is very small, so small in fact that only accurately calibrated measuring instruments can detect the difference in level between front and rear radiation—the ear certainly cannot hear the difference. In brief, then, by selecting a comparatively large piston and judiciously controlling the rear radiation through a *small* measure of acoustic resistance, it is possible to achieve, from the essentially *unbaffled* piston, low-frequency performance equivalent to that obtained from small pistons in sealed boxes.

Because the piston is essentially unbaffled, it performs as a dipole radiator, *i.e.*, each side acts as a radiator of acoustic energy. As a dipole radiator, the spatial distribution of the sound coming from the radiator is a "figure-8" pattern (Fig. 4), that is, there is one lobe of radiation from one side and another lobe of radiation from the other. Such spatial distribution differs considerably from that of a piston in a closed box from which the radiation is generally hemispherical for the low frequencies. For the dipole radiator, then, low-frequency orientation is more directionally positive—which is of special value in stereo reproduction. In conventional boxed systems, the diffusion of the lows throughout the room makes stereo orientation of the lows difficult but with the dipolar directivity of the unbaffled piston, this low-frequency ambiguity is eliminated.

Mechanical Problems

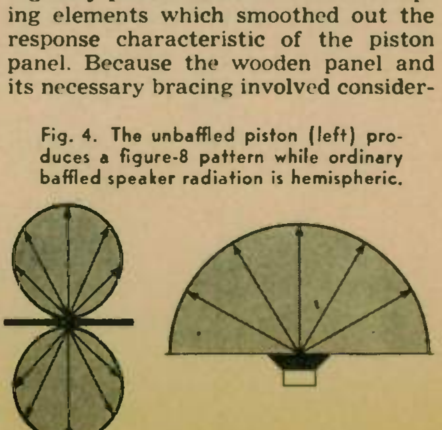
In order to provide the necessary piston action for the large radiation surface and still keep the structure flat, special methods had to be developed to obtain the necessary combination of rigidity and compliance of the panel. This was accomplished by using light, yet rigid, wood sections cemented together and braced in the rear with strategically spaced wooden members. This provides not only the necessary rigidity to the panel but is also effective in adjusting the acoustic response of the system.

Since the entire structure is made of wood, the problem was made more complicated by the fact that a single large panel of wood would not be of uniform density, nor of uniform flexure characteristics. In order to overcome this problem, the piston panel is made of five strips of selected woods—each strip having the same density and elasticity. Once these individual strips have been selected for uniformity and density, they are ready for assembly.

The type of glue or cement used is of extreme importance in maintaining vibrational strength of the panel. Simple cabinetry types of glue are not suitable in that they are essentially hard and brittle and would be shattered by the violent (although small) vibrations of the piston. Yet, on the other hand, it is not possible to use very flexible glues that would not shatter since these flexible glues would not provide the proper adhesion of the panels to provide the controlled piston action required. Accordingly, the adhesive material adopted was a specially devised combination of both thermoplastic and thermosetting types.

Once the panel had been assembled and tested, it was found to contain undesirable, although not excessive valleys and peaks. To eliminate these irregularities, it was found necessary to brace the back of the piston with strategically placed struts and mass damping elements which smoothed out the response characteristic of the piston panel. Because the wooden panel and its necessary bracing involved considerable weight (approximately 185 grams) its means of suspension became one of great stiffness, even to resonate at 20 cps. In fact, the suspension turned out to be more of a rigid clamp than a flexible support. To prevent undue mechanical noises from the piston vibrating against its clamp, a soft, compressed rubber seat which grips the piston under pressure was incorporated, allowing it to flex in this tight grip without vibrating against hard, noise-producing surfaces. Actually, this design has eliminated the flexible support of the conventional speaker cone with its attendant edge-resonance effect and structural fragility.

Fig. 4. The unbaffled piston (left) produces a figure-8 pattern while ordinary baffled speaker radiation is hemispheric.



Because the piston is so rigidly supported, there is minimum excursion of the piston as a whole which permits the voice coil, which is attached to the piston, to be practically completely linked with the magnetic gap of the circuit. In this structure, where the voice coil is so completely linked with the gap, high electrical efficiency can be obtained with magnetic non-linearity reduced to a minimum because of the practically immobile large diaphragm. High electrical efficiency invariably leads to high electrical damping, both of which together produce the conditions for excellent transient response characteristics. Moreover, the large (15" x 22") piston itself sees a high radiation resistance which is true radiation damping—effective in producing good transient response.

While the piston is, by itself, a wide-range reproducer, it was felt that seasoned audiophiles would prefer a little more highs so a tweeter was incorporated in the system. The tweeter is a high-efficiency, driver-type unit in which the radiating surface is a molded phenolic dome-type diaphragm. It is only slightly horn-loaded and has a phase-equalizing button for extending the very top range of the system. The network used is a simple high-pass filter. No effort was made to eliminate the high frequencies from the woofer section. This has two effects. First, by routing some of the high-frequency power to the woofer, there is a better balance of level between the tweeter and woofer section and, second, the middle- and high-frequency power radiated by the woofer, which comes from a comparatively large surface, minimizes the pinpoint effect of the usual tweeter radiation.

Decor and Applications

Let us now discuss some of the possibilities of decor and application inherent in this design. Instead of a cloth-covered box, the audiophile can now work with a furniture-finished, panelled structure which is decorative from both sides and whose sound is not radiated through, or impeded by, a grille cloth. Because it is so thin (4½"), it may be hung on the wall very much as a picture might be. In fact, there have been a number of inquiries regarding the possibility of painting a picture on the surface of the panel so that it would be both a sound radiator and a picture.

For the present, however, we must recognize that in order to maintain the stability of the diaphragm against the ravages of humidity and weathering, the surface of the panel had to be treated as a piece of fine furniture. If one wanted to paint a picture on a finely finished piece of wood which has been sanded, filled, varnished, and buffed, then he is at liberty to do so, but only to the extent that the original treatment of the panel is not destroyed.

The system can also serve as a room divider, either by itself or standing on ready-made metal-shelved room dividers or suspended between floor-to-ceiling pressure posts. The units may also be used as "sound chandeliers" with, perhaps, a lighting source between them and the ceiling so that indirect lighting is obtained while, at the same time, sound is produced directly downward and simultaneously scattered from the back of the speaker up to the ceiling and back into the room.

Another application would be to mount the speakers within the joists of the wall so that the speaker is flush with the wall. Since the speaker is only 4½ inches wide, it may actually be used to radiate from both directions into adjoining rooms. If it is desired to radiate only into a single room, then the back of the speaker simply transmits the sound into the large cavity up and down between the joists and the wall structure.

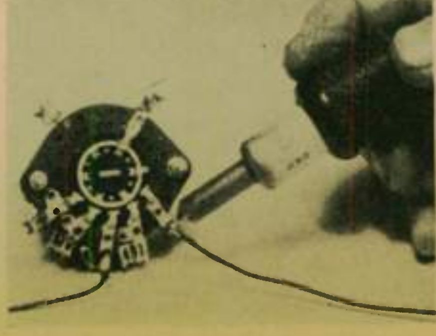
It is also possible to employ the speaker as the actual walls of equipment cabinets, thus saving considerable space (and expensive woodworking) over the original cabinet structure. One unusual application of this speaker involves its installation into the headboards of twin beds, providing stereo reproduction in the bedroom!

SOLDER DOWNWARD!

By ROBERT HERTZBERG

SOME very mysterious difficulties with rotary switches can be traced to disregard of the laws of gravity on the part of the builder when he is soldering wires to the lugs. Molten solder runs like water, and only a tiny globule of it getting into the switch contacts is usually enough to close them tight.

The correct technique is to turn the equipment so that the lug being soldered points downward. Excess solder and flux then both form harmlessly on the wire.



EXTENDED
Stereo System

Method employed in Zenith's 1960 models to control and extend the apparent separation in stereo reproduction.

EDITOR'S NOTE: *Some of our readers may argue that the technique to be described below is an artificial method of producing an exaggerated stereo effect. Others will claim that since stereo itself is an illusion, then any method employed to enhance the illusion is justified. In any case, the system is claimed to be particularly effective when left and right speaker systems are contained in the same enclosure. We feel sure that our readers will be interested in the circuit and method used.*

THE following circuit is used in *Zenith's* new stereo models in order to enhance the stereo effect. According to the manufacturer, the system eliminates the need for the usual small area center listening position and permits the user to control the degree of apparent separation produced. The method is said to be particularly useful in cases where all the speakers are contained in a single enclosure. As will be seen below, a sum and difference technique is used. By controlling the amount of the difference channel various degrees of the stereo effect can be produced.

The block diagram in Fig. 1 shows the basic principle of operation. The stereo cartridge supplies separate left (L) and right (R) channel signals. These signals are then applied through an automatic balance control circuit to a mixer preamplifier. Out of the mixer two types of information are obtained. These are the sum signal (L + R), which is applied to the sum amplifier, and the difference signal (L−R), which is applied to the difference amplifier. A stereo control in the difference channel determines the degree of stereo effect produced. Outputs from both amplifiers are applied to a transformer matrixing circuit. Here the information is separated and applied to the two sets of speakers used.

Mixer Preamp Circuit

The controlling circuit governing the degree of stereo is the mixer preamp shown in Fig. 2. For the sake of convenience assume that the signals coming from the stereo cartridge are −L and −R. These signals are applied to the wiper arms of two 3-meg. potentiometers that are ganged together. These pots are in the grid circuits of the 12AZ7 twin-triode. The controls are so wired that when the signal is increased at one grid, it is decreased at the other grid, and *vice versa*. In this manner, satisfactory balance can be obtained. The twin-triode acts as a preamp as well as a mixer so that we can obtain the sum and difference information.

With a −L signal supplied by the cartridge, there will be +L at the plate of the left preamp because of phase reversal of the tube. If we assume that the left preamp has a gain of 2, then there will be +2L at the plate. This 2L signal is applied through the 68,000-ohm resistor and the .1-μf. capacitor to the output. Also at the plate of the left preamp is a voltage divider consisting of a 15,000-ohm resistor and a 6800-ohm resistor. At the junction of these two resistors there will be some +L signal.

At the same time, −R information is being supplied to the element of the balance control that feeds the other section of the 12AZ7. Also being applied to this same element of the balance control is the L signal from the other section of the 12AZ7. Hence a composite signal, L−R, is impressed on the grid of the lower section of the mixer stage. This signal is also phase inverted by the tube so that it now becomes an (R−L) signal.

The (R−L) information is channelled into two paths. Part of this information is fed through the .022-μf. capacitor to the stereo control, which is basically a gain control for the (R−L) difference channel. Also part of the information is fed through the 120,000-ohm resistor and the .047-μf. capacitor and mixes with the 2L signal from the left preamp. At this point the (R−L) signal combines with the 2L signal to produce an (R+L) sum signal that is fed to the sum amplifier.

Stereo Control & Matrixing

For monophonic operation, the stereo control is at its minimum setting. Under this condition no difference signal is applied to the difference channel. All information that passes through the system is (L+R) which is subsequently channelled to both speaker systems.

For standard stereo operation, the stereo control is advanced until the sum amplifier and difference amplifier receive equal amplitude signals. Under these conditions, (L+R) and (L−R) information is supplied to the matrix and only the L information is applied to the left speaker and only the R information is applied to the right speak-

Fig. 1. Block diagram of arrangement used for "extended stereo."

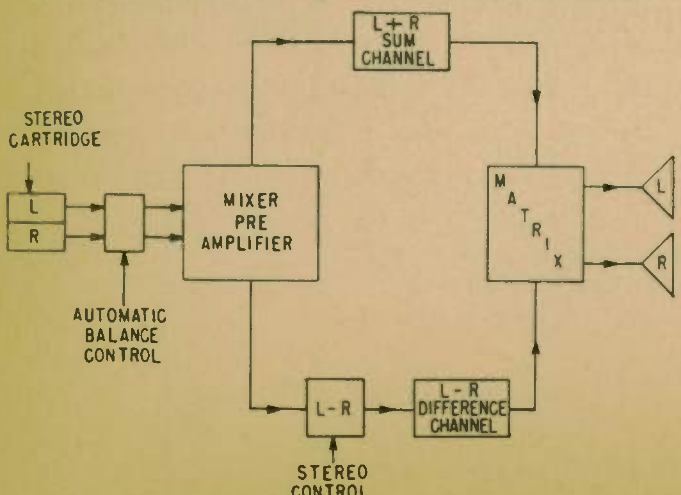
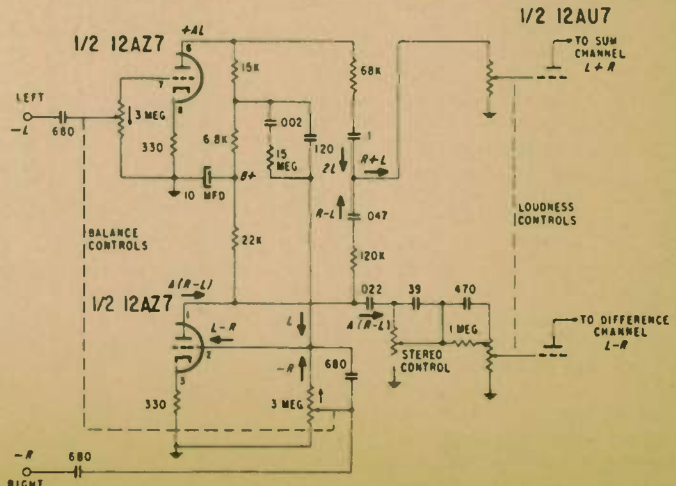


Fig. 2. The mixer-preamplifier circuit employed in Zenith sets.



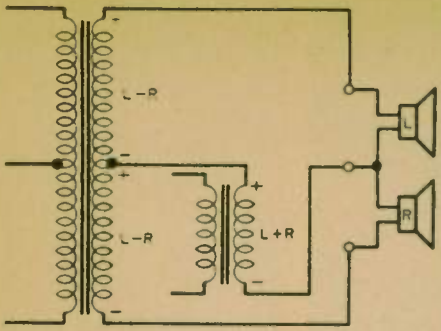


Fig. 3. Basic matrixing circuit used.

er. This is normal stereo operation.

For "extended stereo" operation, the stereo control is advanced further so that the (L−R) or difference-channel signal is greater than the signal applied to the sum (L+R) channel. This difference in relative level is maintained at different loudness settings since the loudness control for both sum and difference amplifiers are ganged and the gains of both channels are controlled simultaneously.

Output matrixing, or mixing of the sum and difference signals in order to get the desired outputs for the speakers, is accomplished by the output transformers (Fig. 3). The secondary winding of the difference-channel (L−R) transformer is center-tapped. To this center-tapped secondary is connected the untapped secondary winding of the sum-channel (L+R) transformer. We now have two windings that are connected in series-aiding (the sum winding and the top half of the difference winding) or in series-opposing (the sum winding and the bottom half of the difference winding). There are three possible conditions of operation as follows:

1. For monophonic operation, there is no difference signal so that both speakers will receive the sum signal.

2. For standard or normal stereo operation, the output of the difference channel is equal to the output of the sum channel. Under these conditions the upper speaker has applied to it (L+R) + (L−R), or 2L. Note that this is the left information only. The lower speaker has applied to it (L+R) − (L−R), or 2R. This is the right information only.

3. For "extended stereo" operation, the output of the difference channel is greater than the sum channel. Assuming there is twice the signal in the difference channel, or 2(L−R), the signal applied to the upper speaker is (L+R) + 2(L−R), or 3L−R. The signal applied to the lower speaker is (L+R) − 2(L−R), or 3R−L. Notice that the left speaker system receives an out-of-phase component of right information along with the left information, and the right speaker system receives an out-of-phase component of left information along with the right information. The magnitude of the out-of-phase components is what determines the apparent spread of the "extended stereo" system.

Here's all you need to build HEATHKIT® HI-FI

...these tools and NO EXPERIENCE

We say "NO ELECTRONIC EXPERIENCE NEEDED" . . . and we mean it. Recognized by leading consumer research organizations as the best in the field of kit construction manuals, the Heathkit step at a time instructions leave absolutely nothing to question. Giant size pictorial diagrams and everyday, non-technical language make every Heathkit easy for anyone to build. No electronic or soldering experience is required; the Heathkit manuals teach you the right way to do a professional job the first time. No matter what your field of interest, building a Heathkit will be one of your most exciting, rewarding experiences.

And you'll save 50% or more on the world's finest electronic equipment. Send today for your *free* Heathkit catalog—or see your nearest authorized Heathkit dealer.

EVERYTHING FOR HI-FI

AMPLIFIERS

TUNERS

CHANGERS

TAPE DECKS

Mail the coupon Today!

HEATHKIT

HEATH COMPANY

a subsidiary of DAYSTROM, INCORPORATED

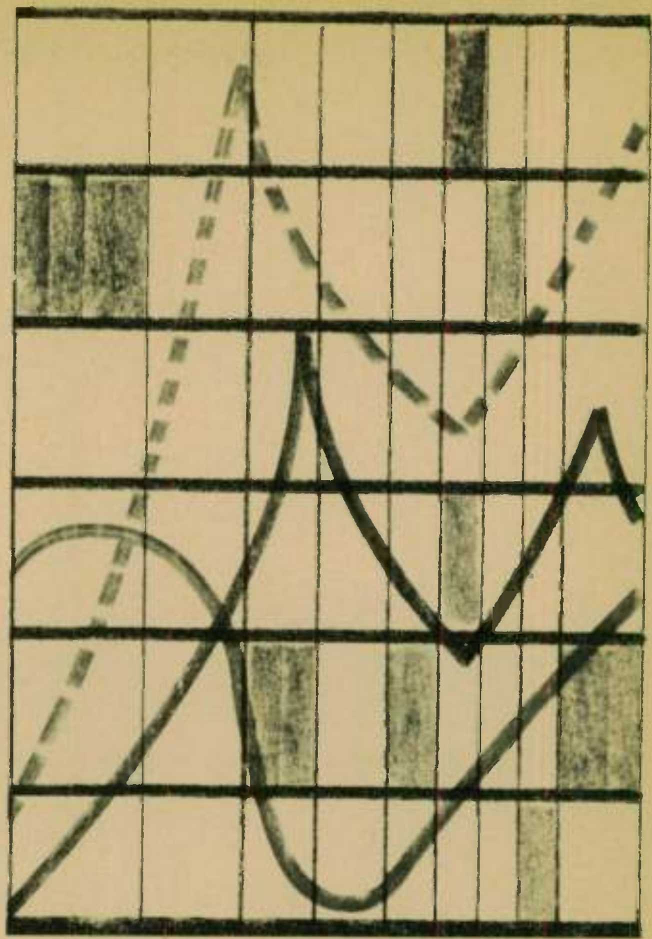
Benton Harbor 15, Michigan

Please send the latest Free Heathkit Catalog.

NAME______________________________

ADDRESS___________________________

CITY_______________ZONE____STATE_______



SECTION · 2

Hi-Fi and Audio Theory and Practice

Finding Faults In Hi-Fi Systems 38

Simplified Speaker Testing 41

Intermodulation Distortion 45

The Decibel Without Pain 48

Finding Faults in Hi-Fi Systems

Comprehensive catalogue of defects, their symptoms, and their probable causes in home audio equipment.

By **DAVID FIDELMAN**

Author, "Servicing Hi-Fi Systems"

THE REQUIREMENTS of high-fidelity reproduction are such that the equipment involved, if it is honestly made, generally will be of better-than-average quality in more ways than one. For example, it is likely to stand up relatively well under regular use. However, any electronic equipment is eventually subject to failure. When trouble crops up in a hi-fi system, understanding the types of failures that occur and their probable effects on over-all operation can greatly simplify servicing.

A complete hi-fi system is a complex chain consisting of many different components or sections. While difficulty can arise in any one section, the problems associated with each — fortunately — tend to be distinctive.

In any system, there are two general methods of locating a failure. One is to trace through the entire system to determine the section in which trouble lies, and then to conduct detailed examinations within this area until the specific trouble point is reached. The other method is to attempt recognition of the section where that trouble is probably occurring from the symptom—through experience or logic, or both. It may then be possible to proceed directly to the area of the defect, sometimes directly to the fault itself.

Of the two methods the second, although it is less "scientific," can be much faster when proper diagnosis is made. Hence, it is useful to compile and classify defects and their symptoms. Assuming normal initial operation of the system, subsequent failures rather than basic system defects will be considered here.

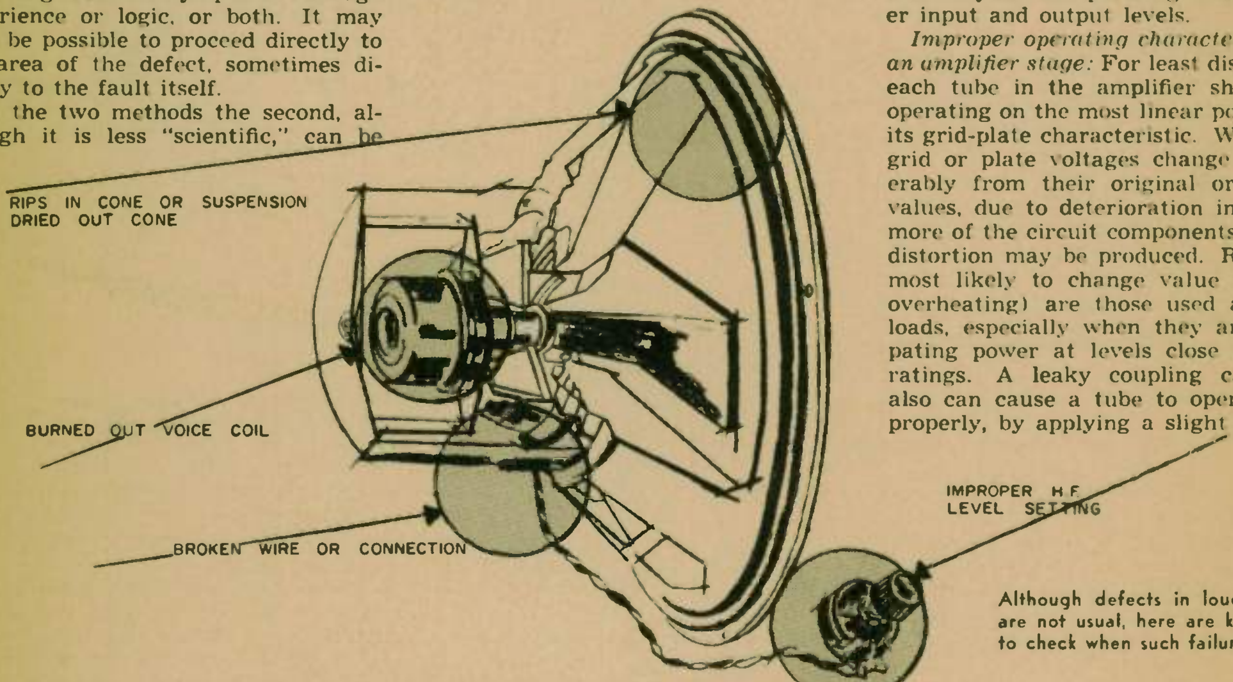
Failures in the Amplifier

Improper operation of the amplifier or preamplifier (generally, as indicated by excessive distortion) can be caused by misadjustments or changes in the circuit due to some component's degeneration, rather than from outright failure. Classes of such failures include:

Overloading. A properly designed amplifier should overload only in the output stage of the power amplifier beyond its rated output. Overloading is easily checked, since reducing the signal level will cause the distortion to be reduced. When the overload occurs below the rated output level, or obviously below a reasonable volume level, component failure in the output stage is probable. If the overloading occurs in some intermediate stage in the circuit, the most likely cause is a tube or component failure that results in a reduced load resistance across some part of the circuit, causing it to overload at a lower level than normal. In many cases the overloading is due to old or weak tubes, and clears up when the bad tube or tubes are replaced. Although any tube in the circuit can produce overloading, the fault usually occurs in driver or power-amplifier tubes.

A failure in the power supply can cause it to produce incorrect voltages, so that tubes may be operating at reduced plate voltages and thus overload at lower than normal levels. It is also possible, by improper operation of the equipment, to produce overloading in a system that has no failures. For example, if the gains and levels of the different units of the system have been misadjusted, it is possible to feed too high a level from one unit into a low-level input of the following unit and thus overload the input stages of the latter. This is easily recognized and corrected by re-adjusting the controls of the different units of the system so that they are all operating at the proper input and output levels.

Improper operating characteristic of an amplifier stage: For least distortion, each tube in the amplifier should be operating on the most linear portion of its grid-plate characteristic. When the grid or plate voltages change considerably from their original or design values, due to deterioration in one or more of the circuit components, severe distortion may be produced. Resistors most likely to change value (due to overheating) are those used as plate loads, especially when they are dissipating power at levels close to their ratings. A leaky coupling capacitor also can cause a tube to operate improperly, by applying a slight positive



Although defects in loudspeakers are not usual, here are key points to check when such failures occur.

voltage to the grid. Incorrect power-supply voltages may also be suspected.

Improper feedback can be a cause of poor amplifier performance. If the feedback circuit or one of the circuits within the feedback loop (or loops) has changed or failed, the result may be parasitic oscillation due to positive feedback, or increased distortion and poor frequency response due to the removal of adequate feedback. The components most likely to be involved are the tubes, resistors and capacitors in the feedback network, and any parasitic-suppression resistors that may be used in the power-amplifier stage. It may sometimes be necessary to change the feedback or the phase-correction network to other than the design values in order to stop a case of high-frequency oscillation. Generally, reducing the amount of feedback will help.

High noise level can be caused by aging or failure of some component. Defective tubes may be the cause of thermal noise, hiss, and crackle from intermittent contacts within the tube, as well as microphonics in low-level stages due to sound pickup from vibration of the tube elements or through direct sound conduction through the mounting of the tube socket. Tubes may also cause noise because of bad contact between the tube base pins and the socket, bad grid-cap contact (if one is used), and intermittent contact between a shield and ground.

Defective resistors and capacitors can produce thermal noise (especially when carbon resistors are used in the plate circuits of low-level stages) and intermittents (for example, voltage breakdown and recovery in the dielectric of a capacitor). Volume controls and switches can be a major source of noise—loose or dirty contacts that produce momentary open circuits are the most common causes of noisy controls or switches.

Hum that is higher than normal can be caused by a failure in one or more of the following components or circuits: power-supply filter capacitor or choke; cathode bypass capacitor; tube heater-to-cathode leakage or short circuit; unbalance in the push-pull power output stage; failure or unbalance in the d.c. biasing circuit (often associated with the heater circuitry); or a poor connection in the shielding or grounding in low-level stages.

Motorboating is a very low-frequency oscillation, due generally to coupling between high-level and low-level stages through the common impedance of the power supply. It is most frequently the result of defective electrolytic capacitors or decoupling networks, although overloading of the output transformer or unbalance in the push-pull output stage can also cause coupling through the power supply.

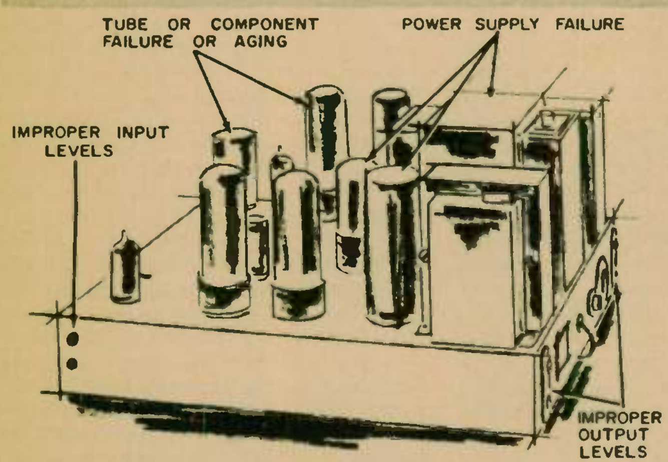
Phonograph Failures

The phonograph can be a frequent source of trouble, and its performance has become even more important with improvements in records. Improper operation can cause either complete or partial loss of performance in the record-reproducing channel, introduce severe distortion, and even cause irreparable damage to valuable records if not corrected in time. Some of the most common problems are noted here.

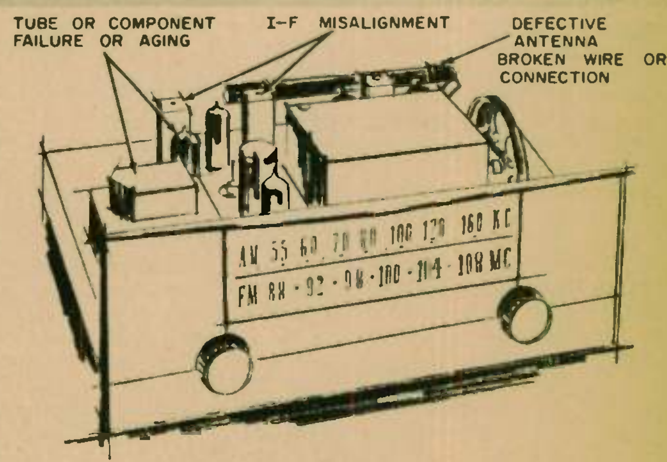
A worn or chipped stylus is one of the first hazards in the case of the phonograph. As this element wears, flats begin to form on the sides so that the stylus eventually assumes the shape of a chisel with sharp edges and it begins to ride in the bottom of the groove instead of gliding along the sides. On monophonic, microgroove records, this wear can be quite serious after only a few playings with a metal-tipped stylus, fewer than a hundred playings with a sapphire stylus. Where a stylus is sufficiently worn, it is even possible to ruin a long-playing record with a single playing. This hazard is considerably more serious in the case of stereophonic records.

Stylus-assembly defects are generally, although not always, the result of careless handling or other abuse of the record-playing equipment. There are a number of ways in which this assembly may become defective. It may be loosely mounted or otherwise not properly coupled to the cartridge. The stylus may not be making proper contact with the record groove due to improper positioning of this assembly, or because it is bent out of shape, or because one of the pole pieces is bent or out of position (in the case of a magnetic cartridge). Stylus motion may be impeded because dust or other foreign matter has accumulated in the assembly, or against such associated portions of the cartridge as the guard or pole pieces.

Other possible failures include difficulty with the cartridge itself. A crystal or ceramic element may become broken or otherwise defective. In a magnetic cartridge, coil wires may become open or shorted, or the magnetic circuit itself may become defective. A tone arm may be damaged or improperly adjusted. There may also be some binding in its bearings or mounting, restricting movement. High hum level, intermittent noises, or intermittent signal may indicate a loose or broken cartridge lead wire.



Possible failures and check-points in the basic amplifier.



Some things to look for in a tuner that is giving trouble.

Turntable rotation failure can occur either in the motor or in the drive mechanism from the motor to the turntable. Causes may be: lack of power to the motor due to a failure in the electrical circuit (such as a defective switch or plug, or a break in the wiring); wiring failure in the motor; mechanical failure in the motor (for example, binding due to damaged or frozen bearings, or to gummed oil or foreign material between the armature and pole-piece); failure in the drive mechanism (for example, worn idler-wheel bearings or rubber tires; stretched rubber drive belts, insufficient friction between rubber and metal parts due to oil or grease, insufficient spring tensions); binding in the turntable shaft or defective bearings.

Trip-cycle defects are generally due to improper adjustments, binding in release or friction components, defective springs or bent parts. The electrical circuit may also be at fault here (defective trip switches and solenoids).

tute distortion. But when these frequencies are not harmonically related, then the audible effects are quite unpleasant.

Limits of IM

In measuring IM distortion it is common to apply to the equipment under test a large low-frequency signal along with a small high-frequency signal. The usual ratio between the two amplitudes is 4 to 1 (12 db). Low frequencies in common use are 40 cps, 60 cps, 100 cps, and 150 cps; high frequencies in common use are 3000 cps, 4000 cps, 6000 cps, and 15,000 cps.[1] Depending on which frequencies are used, and their related amplitudes, different IM distortion figures will be obtained. Therefore, it is important to state just what frequencies and amplitudes are used in a given test when specs are quoted. It is also important to state the power level at which the measurement is made.

The amount of IM distortion that should be present in top-quality hi-fi equipment has been the subject of much discussion. According to one authority[2] the following arbitrary grouping is given:

AMPLIFIER	IM
Very high-fidelity (lower test freq., 40 cps)	Less than 2%
High-fidelity (lower test freq., 40 cps)	Less than 4%
Good fidelity (lower test freq., 60 cps)	Less than 8%
Fairly good fidelity (lower test freq., 60 cps)	Less than 20%

Recently the *Heath Co.* has set up its own standards for IM distortion limits on power amplifiers. Its groupings, using test frequencies of 60 and 6000 cps at a 4:1 ratio, are as follows:

AMPLIFIER	IM
Professional	1% or less
High-fidelity	2% or less
Utility	3% or less

In our own lab-tested reports as well as in the specifications quoted by a large number of companies in the field, an IM distortion figure of 1 per-cent is used. If we find that an amplifier will not produce its full rated output power at 1 per-cent IM, then we down-rate the unit. Admittedly this is an extremely strict specification to meet but we feel that it is fair provided all units are subjected to the same criterion.

IM Compared with THD

In general, when an amplifier has low intermodulation distortion, it also has low total harmonic distortion (THD). When the IM is high, so is the THD. However, it must be remembered that these two methods of measuring distortion are quite different, so it is logical to expect that the percentage figures for IM and THD will not be the same.

The relation between IM and THD in a hypothetical amplifier is shown in Fig. 3. These curves are fairly typical. Note that at low power outputs (up to about 11 watts in this case) the IM distortion is less than the THD, as measured at the extremes of the frequency range to be covered (say 20 or 20,000 cps). At 10 watts output, the IM of this particular amplifier is 1% while the THD is 2%. It is quite accidental that these two figures, which we use as our own standards, occur at the same power level. But observe what happens at the higher power levels. First, the IM goes upward a little more suddenly than does THD. For this reason, many feel that the IM measurement is a much more sensitive indication of amplifier overload. Second, the two distortion curves cross each other so that the value of the IM becomes higher than the value of the THD.

If the harmonic distortion measurements are made at the mid-frequencies (400 or 1000 cps), our amplifier is seen to produce much less THD. Under these conditions, it is not uncommon for the value of THD to be less than the value of IM at all power levels. In the case of our hypothetical amplifier, an output power of 13 watts is produced at a THD of 2 per-cent. At this power level, the IM is close to 8 per-cent, or almost 4 times as great.

It is quite possible to have a totally different relation between IM and THD than is shown in Fig. 3. The design of the amplifier being checked, and particularly how it overloads, may change this picture markedly.

IM Analyzers

Although IM distortion can be measured with separate audio generators, filter circuits, oscilloscope or meter indicators, common practice is to combine many or even all the functions needed into a single instrument. This instrument may be called an IM analyzer. A block diagram of all the functions performed by such an instrument is shown in Fig. 2.

Two stable audio generators are used, one of these produces the low frequency and the other produces the high frequency required for the test. In some instruments these frequencies are adjustable, in others they are fixed. The two signals are then fed into attenuators for isolation and for adjustment of the amplitude ratio (usually 4 to 1). Some units permit the user to

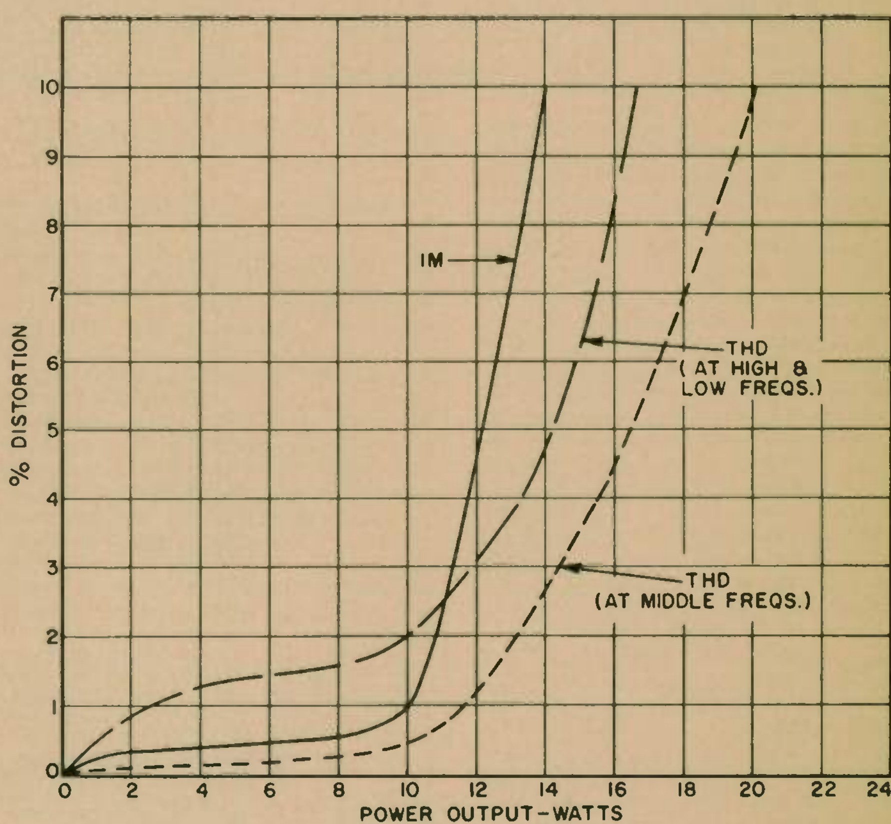
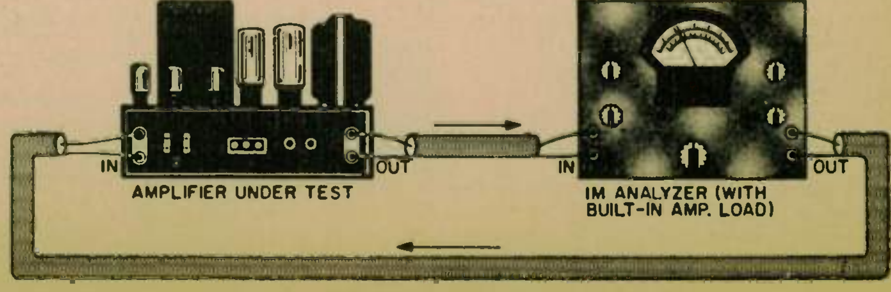


Fig. 3. Relation between IM and total harmonic distortion in hypothetical amplifier.

Fig. 4. Test setup employed for checking IM distortion of a high-fidelity amplifier.



[1] This method is the one specified by the Society of Motion Picture & Television Engineers.
[2] "Radiotron Designer's Handbook" Fourth Edition—edited by Langford-Smith.

voltage to the grid. Incorrect power-supply voltages may also be suspected.

Improper feedback can be a cause of poor amplifier performance. If the feedback circuit or one of the circuits within the feedback loop (or loops) has changed or failed, the result may be parasitic oscillation due to positive feedback, or increased distortion and poor frequency response due to the removal of adequate feedback. The components most likely to be involved are the tubes, resistors and capacitors in the feedback network, and any parasitic-suppression resistors that may be used in the power-amplifier stage. It may sometimes be necessary to change the feedback or the phase-correction network to other than the design values in order to stop a case of high-frequency oscillation. Generally, reducing the amount of feedback will help.

High noise level can be caused by aging or failure of some component. heater-to-cathode leakage or short circuit; unbalance in the push-pull power output stage; failure or unbalance in the d.c. biasing circuit (often associated with the heater circuitry); or a poor connection in the shielding or grounding in low-level stages.

Motorboating is a very low-frequency oscillation, due generally to coupling between high-level and low-level stages through the common impedance of the power supply. It is most frequently the result of defective electrolytic capacitors or decoupling networks, although overloading of the output transformer or unbalance in the push-pull output stage can also cause coupling through the power supply.

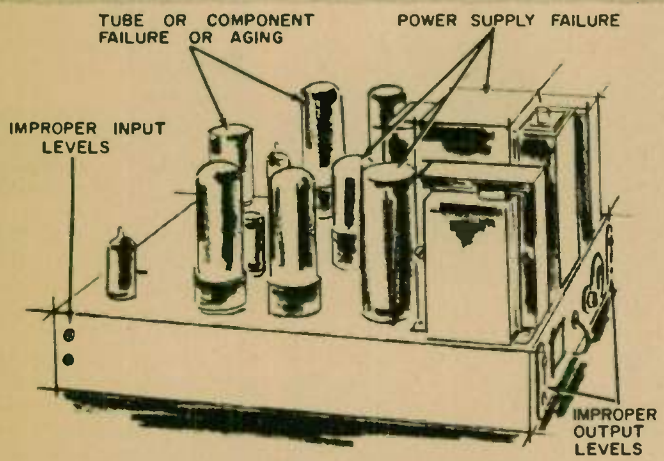
Phonograph Failures

The phonograph can be a frequent source of trouble, and its performance has become even more important with improvements in records. Improper operation can cause either complete or partial loss of performance in the record-reproducing channel, introduce severe distortion, and even cause irreparable damage to valuable records if not corrected in time. Some of the most common problems are noted here.

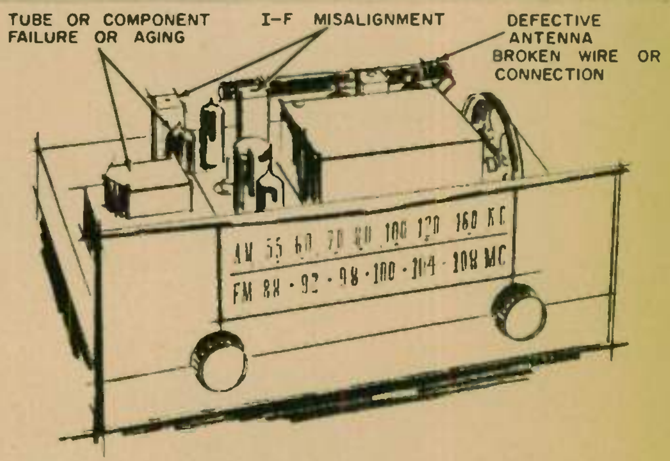
A worn or chipped stylus is one of the first hazards in the case of the phonograph. As this element wears, flats begin to form on the sides so that the stylus eventually assumes the shape of a chisel with sharp edges and it begins to ride in the bottom of the groove instead of gliding along the sides. On monophonic, microgroove records, this wear can be quite serious after only a few playings with a metal-tipped stylus, fewer than a hundred playings with a sapphire stylus. Where a stylus is sufficiently worn, it is even possible to ruin a long-playing record with a single playing. This hazard is considerably more serious in the case of stereophonic records.

Stylus-assembly defects are generally, although not always, the result of careless handling or other abuse of the record-playing equipment. There are a number of ways in which this assembly may become defective. It may be loosely mounted or otherwise not properly coupled to the cartridge. The stylus may not be making proper contact with the record groove due to improper positioning of this assembly, or because it is bent out of shape, or because one of the pole pieces is bent or out of position (in the case of a magnetic cartridge). Stylus motion may be impeded because dust or other foreign matter has accumulated in the assembly, or against such associated portions of the cartridge as the guard or pole pieces.

Other possible failures include difficulty with the cartridge itself. A crystal or ceramic element may become broken or otherwise defective. In a magnetic cartridge, coil wires may become open or shorted, or the magnetic circuit itself may become defective. A tone arm may be damaged or improperly adjusted. There may also be some binding in its bearings or mounting, restricting movement. High hum level, intermittent noises, or intermittent signal may indicate a loose or broken cartridge lead wire.



Possible failures and check-points in the basic amplifier.



Some things to look for in a tuner that is giving trouble.

Defective tubes may be the cause of thermal noise, hiss, and crackle from intermittent contacts within the tube, as well as microphonics in low-level stages due to sound pickup from vibration of the tube elements or through direct sound conduction through the mounting of the tube socket. Tubes may also cause noise because of bad contact between the tube base pins and the socket, bad grid-cap contact (if one is used), and intermittent contact between a shield and ground.

Defective resistors and capacitors can produce thermal noise (especially when carbon resistors are used in the plate circuits of low-level stages) and intermittents (for example, voltage breakdown and recovery in the dielectric of a capacitor). Volume controls and switches can be a major source of noise—loose or dirty contacts that produce momentary open circuits are the most common causes of noisy controls or switches.

Hum that is higher than normal can be caused by a failure in one or more of the following components or circuits: power-supply filter capacitor or choke; cathode bypass capacitor; tube

Turntable rotation failure can occur either in the motor or in the drive mechanism from the motor to the turntable. Causes may be: lack of power to the motor due to a failure in the electrical circuit (such as a defective switch or plug, or a break in the wiring); wiring failure in the motor; mechanical failure in the motor (for example, binding due to damaged or frozen bearings, or to gummed oil or foreign material between the armature and pole-piece); failure in the drive mechanism (for example, worn idler-wheel bearings or rubber tires; stretched rubber drive belts, insufficient friction between rubber and metal parts due to oil or grease, insufficient spring tensions); binding in the turntable shaft or defective bearings.

Trip-cycle defects are generally due to improper adjustments, binding in release or friction components, defective springs or bent parts. The electrical circuit may also be at fault here (defective trip switches and solenoids).

Other causes may be in the records themselves, which may be of incorrect size, warped, damaged, have an oversized center hole, or non-standard lead-in and run-out grooves.

Record-dropping problems are also often the fault of the records. They can be caused by records that have broken or chipped edges, that are warped, or that are too thick or too thin. Defects in the changer that can cause record-dropping problems include: bent spindle assemblies or record-selector posts; sticking or binding of sliding parts in the record selecting or push-off mechanism; bent pusher shafts; loose or improperly set push-off arms and cams.

A number of other failures can occur in record-player mechanisms, such as: wow and flutter, rumble, mechanical resonances in the tone arm or other structures, and 60-cycle or 120-cycle hum due to electromagnetic pickup. Where these are due to failures in the record-player mechanism, such as improper lubrication of the motor and mechanism, flats or binding of the idler wheel, dirty or defective bearings, grease on the idler wheel or turntable rim, loosened screws, poor electrical connections, and other such defects that have arisen in the course of normal operation of the unit, they can be corrected and eliminated by proper servicing and maintenance. However, in many cases, these distortions and defects are partly due to the basic design of the record player, and are not actually failures in the unit.

Loudspeaker Troubles

A loudspeaker properly matched to the system will rarely fail or wear out in normal use. Where such failures do occur they are most often the result of some sort of abuse to the loudspeaker. These failures include:

Burned-out voice coils due to extreme overloading caused by connecting the loudspeaker to an amplifier delivering much more power than the rating of the loudspeaker, or by accidentally connecting the 117-volt a.c. line across the voice coil.

Rips and tears in the cone or suspension are due to careless handling or insufficient precautions to protect the speaker in installation.

Poor frequency response and distortion are often found in woofer-tweeter combinations where one or both speakers has an independent level control; this control can be misadjusted to change the relative levels of the two speakers, thus giving poor frequency response and possible distortion.

Rattles are due to loose (or broken) pieces in the loudspeaker enclosure. Resonances may occur if some of the stiffening braces have come loose from the walls of the enclosure (for example, due to aging and loosening of the glue which secures them).

Acoustic feedback or mechanical feedback may develop if the location of different units in the system has been changed in such a manner that the loudspeaker can feed back acoustical or mechanical vibrations to the phonograph pickup cartridge or other low-level sections of the system.

Poor sound conditions may be due to a number of factors that have nothing to do with the electrical operation of the system itself. Improper placement of the loudspeaker system in the room, or a loudspeaker system that is not well matched acoustically to the room can be a cause of poor sound reproduction. Very often, the history of a recent change in the furnishings or decor of the room can be one cause of poor room acoustics, and thus affect the performance of the system.

Tuner Problems

Failures and deteriorations in performance of high-fidelity tuners require a different type of consideration from those which occur in ordinary radio and TV receivers. Since the performance requirements are much more critical, relatively slight deteriorations must be included as equipment failures. Some points to consider follow:

Detector failures in AM tuners can be caused by a change in the diode load, and will show up as a distorted output. The diode load resistor (generally 2 to 10 megohms) may have changed in value, or the detector or the following tube may have failed. If the tuner uses a cathode-follower type of detector, trouble may result from changes in the cathode resistor or the r.f. bypass capacitor.

Distortion in FM tuners can be caused by misalignment of the i.f. amplifier or the detector, or by regeneration in the former. Such symptoms are sometimes due to a change in the value of an i.f. transformer's damping resistor or to a failure in the bypassing components in addition to direct misalignment of the i.f. transformers.

Poor frequency response may be due to reduced r.f. or i.f. bandwidth. This can happen if any of the damping resistors across the i.f. transformers have opened or increased in value. In FM tuners, poor frequency response can also be caused by a component failure in the audio de-emphasis network.

High noise level in AM tuners may be due to misalignment of the r.f. or i.f. transformers, or to a component failure in the whistle filter (if the tuner has one). In FM receivers, noise is generally due to incomplete limiting, which can be caused by reduced sensitivity due to faulty tubes or components, or to a failure in the antenna system or lead-in wire.

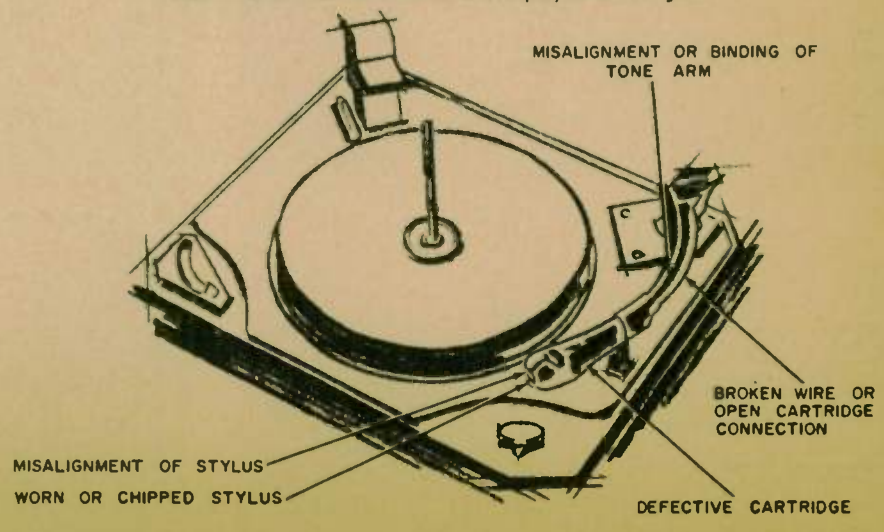
Tuners that contain preamplifiers and control units are also subject to the same failures as amplifiers and the possibility of a failure in the audio section should always be considered.

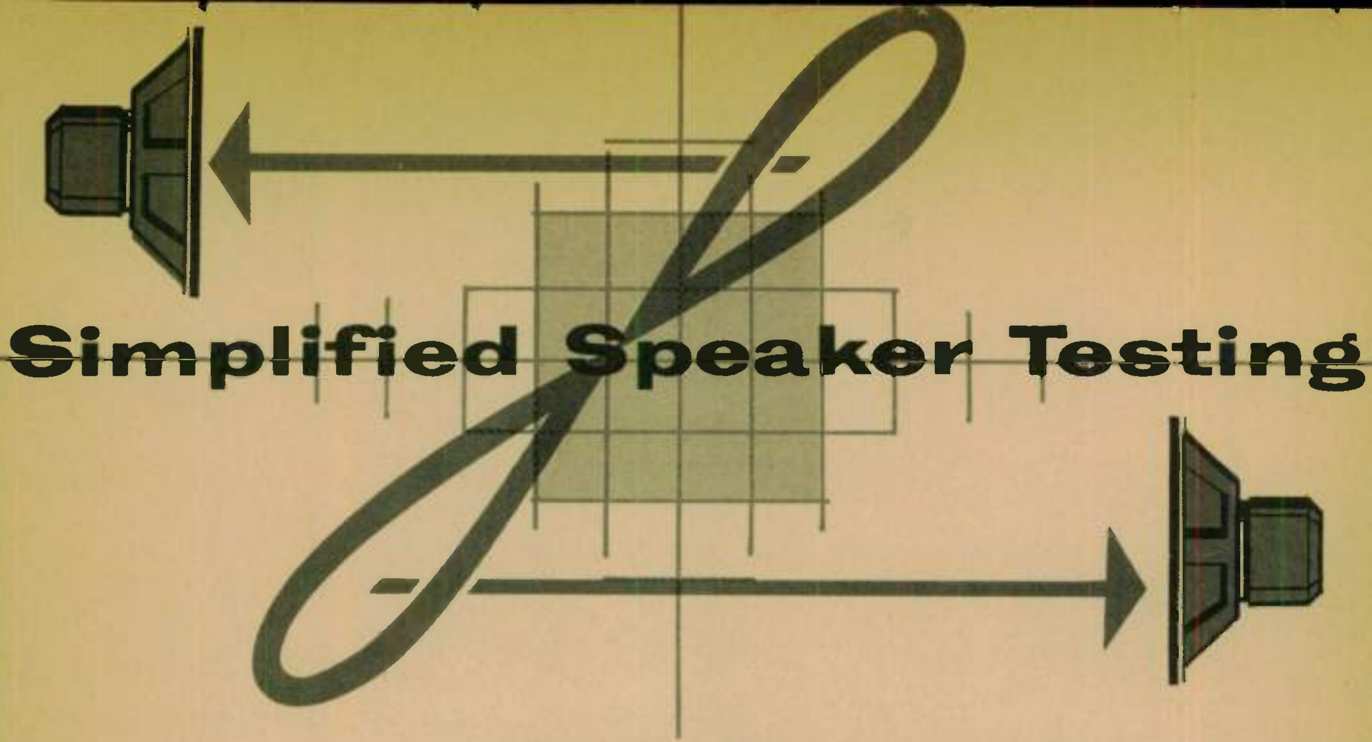
General Problems

It should always be kept in mind that failures in the system set-up itself may occur that are not defects within individual components in the system. Fortunately, these will generally be in the nature of installation errors rather than problems that develop subsequently, but the possibility of this occurrence must be considered. Sometimes such general defects, although they were part of the original installation, do not become annoyingly evident until later. Sometimes it is actually possible that a latent installation defect (a marginally good interconnection, for example) will not cause trouble for a while.

System failures are generally involved with interwiring between components (broken or fraying wires, loose connections, open grounds, ground loops, pulled-out plugs). Changes may take place because various units in the system, or even the entire system, have been moved or re-located in line with a recent attempt to redecorate the room, or even a recent cleaning of the room. For this reason, it always pays to check up on what has been happening in the room where the system is located. However, once a system has been giving satisfactory operation for some time, one would scarcely jump at such possibilities first; a failure within one of the units involved is the greater probability.

Most common faults to occur in record players or changers.





Simplified Speaker Testing

By R. D. HERLOCKER

Use an oscilloscope and audio oscillator to measure loudspeaker impedance, phase angle, and distortion.

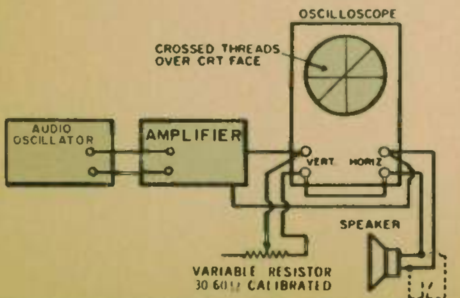
MANY audiophiles test their own amplifiers either as a matter of routine maintenance to keep their equipment in top-notch working order or to modify them for improved operation. On the other hand, it seems that relatively few make any attempt to check the performance of their speaker systems. This is in spite of the well-known fact that the average amplifier of today is freer from distortion and has a more uniform frequency response than even the better speakers now in use.

What accounts for this difference in attitude? Is it because so many of the factors controlling response and distortion are "built-in" to the speaker and thus are presumably not subject to improvement? Or is it because so many persons seem to have the idea that in order to determine any significant characteristics of a speaker system, test equipment is needed which is beyond the reach of all but well (and expensively) equipped laboratories?

It is quite true that many items which affect response and fidelity are inherent in any speaker and are fixed at the time of manufacture. However, other factors, which are also quite important in determining the over-all response of the system, are very much under the control of the user. These controllable factors are primarily those which are concerned with mounting or enclosing the speaker and locating it within the room, and their importance should not be underestimated. For example, a speaker with a free-air resonance near 30 cycles, or even lower, and a smooth response above that point, may sound best in an infinite-baffle type of enclosure, while another speaker with a sharp resonance falling within the normal bass range (50 to 70 cycles, for instance) may need a bass-reflex type enclosure to tame the resonance and generally smooth out the lower end of its range. The type, amount, and location of sound absorbing material within the enclosure will also affect the response, but usually in a higher range, since the lack of such material may permit other resonances due to reflections within the enclosure. Then, too, if tests can be made in the room in which the speaker is to be used, it will probably be found that the location of the speaker in the room has a considerable effect on its performance. This is particularly true in the case of stereo reproduction.

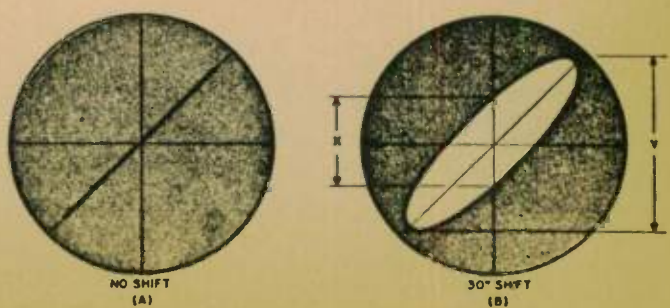
It seems to be commonly felt that any measurements of loudspeaker performance must necessarily be made with a lot of complicated and expensive equipment, which might typically include a microphone which has been calibrated over the frequency range to be tested, an expensive audio oscillator, a couple of amplifiers, distortion and output meters, and other associated pieces of equipment, and preferably including either an anechoic chamber or "wide open spaces" for free-field testing. Rigorous testing under these conditions will undoubtedly give results which are as nearly definitive as can be had and which will correlate well with the sounds as actually heard. Of course, we all recognize that these sounds constitute the ultimate test of any audio system, although in many cases where the sound is considerably less than perfect, tests, even on the small scale which can be made at home, will help to pinpoint the cause or causes of faulty response, and may suggest measures to improve the

Fig. 1. Equipment setup for speaker tests.



If one side of amplifier output is grounded, unground this lead. 10μf CAPACITOR(S) (SEE TEXT)

Fig. 2. Shown here are a pair of distortion-free oscilloscope traces with and without some phase shift.



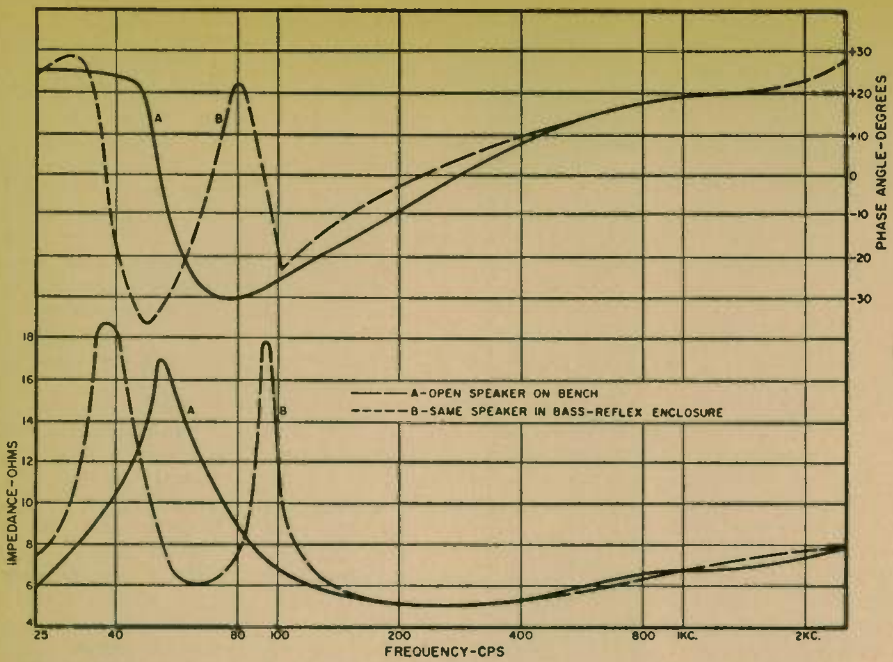


Fig. 3. Typical phase-angle and impedance curves for mounted and unmounted speaker.

over-all performance of the system.

Equipment

Much valuable information about your speaker and its performance can be obtained with simple equipment, such as is available to many experimentally inclined audio fans. The characteristics which can be measured with this simple setup are those which involve the electrical system of the speaker, either directly or because of energy reflected back into it from the mechanical or acoustical systems. They include impedance and phase shift and qualitative indications of harmonic distortion and useful power handling capacity. The equipment needed includes an audio oscillator capable of producing good sine waves over the desired frequency range: 20 to 20,000 cycles should be more than adequate; a power amplifier: ten watts is big enough for most purposes, but it must be free

X/Y = sin ø	ø	X/Y = sin ø	ø
0.00	0°	0.50	30°
0.05	3	0.55	33½
0.10	6	0.60	37
0.15	9	0.65	40½
0.20	12	0.70	44½
0.25	15	0.75	48½
0.30	18	0.80	53
0.35	21	0.85	58
0.40	24	0.90	64
0.45	27	0.95	71
		1.00	90

Table 1. Abridged sine-function table.

from distortion over the frequency range to be measured; an oscilloscope: practically any one will do; and a calibrated wire-wound variable resistor. Thirty ohms maximum resistance is about right for working with 8-ohm speakers and 50 to 60 ohms for 16-ohm speakers. The resistor calibration can be made with a low-range ohmmeter, such as is incorporated in any vacuum-tube voltmeter. This gives sufficient accuracy for present needs, since relative values are more important than absolute values in this application.

The equipment is connected as indicated in Fig. 1. If the oscilloscope does not have a measuring grid over the face of the cathode-ray tube for determining the size of the trace, a satisfactory substitute may be made by simply fastening three threads over the face of the tube with *Scotch* tape. One of these should go vertically across the center of the face and the second horizontally, as determined by the vertical and horizontal traces of the oscilloscope. The third thread is to go across the intersection of the first two, at a 45° angle, from lower left to upper right. For all measurements to be discussed here, the oscilloscope must be adjusted so that the vertical and horizontal gains are equal and the no-signal point is at the intersection of the three threads.

Impedance

Impedance is the simplest characteristic of a speaker to measure. Using the hook-up of Fig. 1, feed a single-frequency signal into the oscilloscope network, adjusting the amplifier output so that the trace nearly fills the face of the cathode-ray tube. Adjust the variable resistor to give a trace that falls on the 45° thread, as in Fig. 2A, if the trace is a single line. If the trace is an ellipse instead of a line, as is more apt to be the case, the proper resistor setting will put the trace with the long axis of the ellipse on the 45° thread, as in Fig. 2B. The impedance of the speaker at the frequency being tested is now the same as the resistance of the variable resistor. Measurements should be repeated at intervals over the desired frequency range and a curve drawn connecting the measured points, as in Fig. 3, for easier visualization of the results. Impedance should be plotted against the logarithm of frequency. Usually a change of 20% to 25% in frequency between adjacent points is permissible, although intermediate points will be needed at frequencies where a small change in frequency produces a disproportionately large change in impedance, as is the case in the vicinity of resonant frequencies.

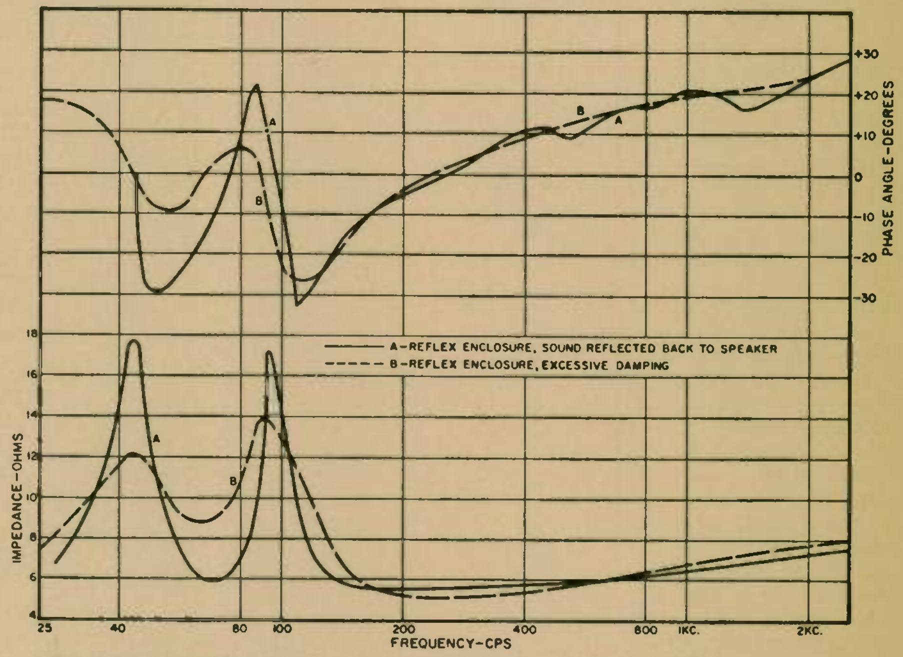


Fig. 4. Phase and impedance curves in reflex cabinet under two different conditions.

Phase Shift

The elliptical trace which will be found throughout a large part of the frequency range of any speaker is a direct reflection of phase shift in the speaker. Phase shift can be caused by reactive components in the electrical, mechanical, or acoustic networks of the speaker and is inherent in these networks. While it can be modified by changes in any of these networks, the acoustic network is the only one which can be readily changed by the home experimenter, since this is the one which is controlled by speaker mounting and by enclosure dimensions and design. Phase shift, in itself, is neither good nor bad in its effect on the sound put out by the speaker system. What is bad is the presence of abrupt changes in phase shift, especially at frequencies differing widely from the bass resonant frequency or frequencies of the system.

The phase shift of a speaker system can be readily determined at the same time the impedance is being measured. Measurements of two dimensions of the oscilloscope trace are needed at each frequency. These are the overall height of the trace and its height through the center (Y and X, respectively, in Fig. 2B). Now $X/Y = \sin \phi$ so the value of the phase angle, ϕ, can be found in any trigonometry table from the ratio between these two heights, or it may be estimated from the abridged sine table given in Table 1.

Whether the reactance causing the phase shift is inductive (positive shift) or capacitative (negative shift) can easily be checked by momentarily placing a capacitor across the speaker terminals, as shown by the dotted lines in Fig. 1. The value of the capacitor is not important, as long as it is large enough to change the shape of the ellipse noticeably when contact is made. A 10 µf. filter capacitor has served quite satisfactorily in the author's tests. If the ellipse is fatter with the capacitor in the circuit, the shift is negative and the reactance of the speaker is capacitive. If the ellipse becomes thinner when the capacitor is placed in the circuit, the shift is positive and the speaker reactance is inductive.

At high frequencies there is normally a positive shift of 20° to 45°, since the inductive reactance of the voice coil is sufficient to have a noticeable effect in this range. At some intermediate frequency, the phase angle will pass through zero and then become negative as the frequency is lowered. Near the resonant frequencies of the speaker and enclosure, the phase angle is apt to undergo quite rapid changes, as shown in Fig. 3. These rapid changes may indicate unevenness of response, particularly if the amplifier with which the speaker is to be used is not highly damped. At frequencies above about 200 cycles, an uneven phase curve is related to the response. For example, the rather erratic phase curve A in Fig. 4 was found to be due to reflections from an enclosure wall, back to the speaker cone. The disturbance was not large enough to show up on the impedance curve. A small change in the enclosure, to eliminate the reflections, smoothed out the curve, as in B of Fig. 3, and at the same time removed a distinct coloration which earlier had been noticed in the sound from this particular speaker-enclosure combination.

The internal damping of a bass-reflex enclosure has quite an effect on the impedance and phase curves obtainable, as well as on the sound emanating from it. This point is illustrated by a comparison of curves B of Figs. 3 and 4, which were taken with the same speaker-enclosure combination. The only difference between the two sets of conditions was that, for curves B of Fig. 4, additional damping material was used, including cotton sheeting across the port to provide extra acoustical resistance at that spot. Curves B of Fig. 4 can be seen to be much smoother, without the sharp impedance peaks or sudden phase changes at resonant frequencies noted in curves B of Fig. 3. However, a price must be paid for all this, which is not apparent from examination of the curves. The low end was quite smooth sounding, but very noticeably weakened by the additional damping. The moral is that test results such as these should always be interpreted in conjunction with actual listening.

Distortion

Harmonic distortion developed in the speaker is made apparent in this test procedure by deviations of the oscilloscope trace from the straight line or elliptical shapes previously discussed. It is in this connection that a distortion-free amplifier between the audio oscillator and the speaker is especially important, since a distorted signal being fed to the speaker will complicate the trace on the oscilloscope and may easily make it seem that the speaker is distorting the signal when, in reality, the amplifier or even the generator, may be responsible for the distortion.

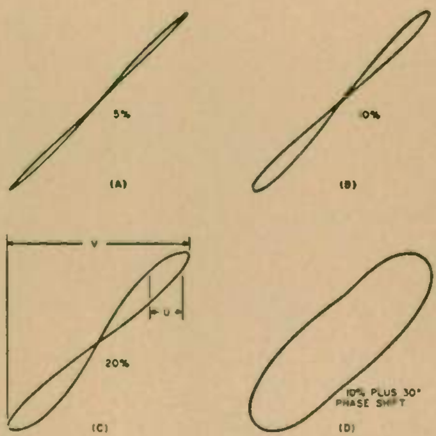
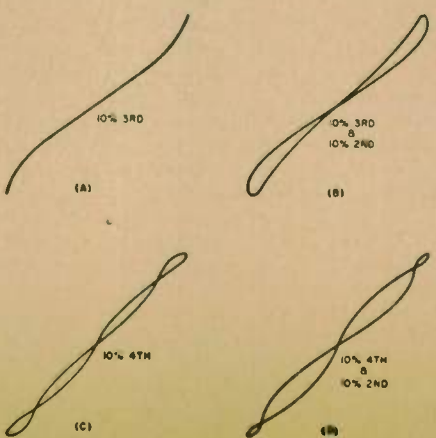


Fig. 5. Traces of 2nd harmonic distortion.

Fig. 6. Oscilloscope traces that contain both 3rd and 4th harmonic distortion.



Accurate measurements of harmonic distortion are not feasible by this method although, with care, a fairly good estimate may be had. In the useful audio-frequency range, by far the commonest harmonic distortion met with is second harmonic, otherwise known as doubling, in the lower frequency range. Figs. 5A, B, and C show the traces obtained with 5%, 10%, and 20% second harmonic added to the fundamental frequency without any phase shift in the speaker. Fig. 5D shows a trace containing 10% second harmonic, but with a 30° phase shift of the fundamental. It is obviously almost impossible to get any valid estimate of harmonic distortion from a trace of this type, so to be able to estimate the distortion, the phase shift must be compensated for. This can be done, at least approximately, by shunting either the variable resistor or the speaker with a suitable capacitor, until the phase shift is eliminated on the oscilloscope trace. The value of the capacitor will have to be found by trial, since it is dependent on frequency, the impedance being shunted, and the phase shift to be compensated. The capacitor should be placed across the speaker, as in Fig. 1, to compensate for a positive (inductive) phase shift or across the resistor to compensate for a negative (capacitive) shift. When second harmonic distortion alone is present, it may be estimated as follows:

% Second Harmonic $= 100\ (U/V)$

where U is the width of the widest part of the curve and V is the overall width of the trace (see Fig. 5C).

Third and higher harmonics will be found much less often. Their estimation does not seem to be practical with this simplified testing setup. Examples of curves containing third and fourth harmonics, alone and in combination with second harmonics, are given in Fig. 6, so that they can be identified, even though they cannot be measured. Third harmonic distortion (no phase shift) gives a curved line, while the fourth harmonic alone gives a triple crossing loop. It can be seen from examination of Figs. 5 and 6 that analysis of the traces to give distortion values is feasible only in the simplest cases.

Some indication of the power handling capacity of a speaker can be had by repeating the tests, particularly in the low-frequency range (say below 200 cycles) at successively higher powers, cutting back the oscilloscope gain as power is increased to keep the trace on the face of the tube. At any given frequency the voltage across the speaker and its impedance can be measured at the power level at which the shape of the trace becomes markedly distorted. Volts squared divided by impedance equals watts (power), which in this case is the useful power capacity of the speaker. Two notes of warning on this kind of testing should be sounded: First, stay within the manufacturer's power ratings on both the speaker and the variable resistor to avoid physical damage to them. Second, a continuous tone, such as is produced in these tests, can seem intolerably loud even at a power input of only one to two watts. This last statement applies with double force to uninterested wives.

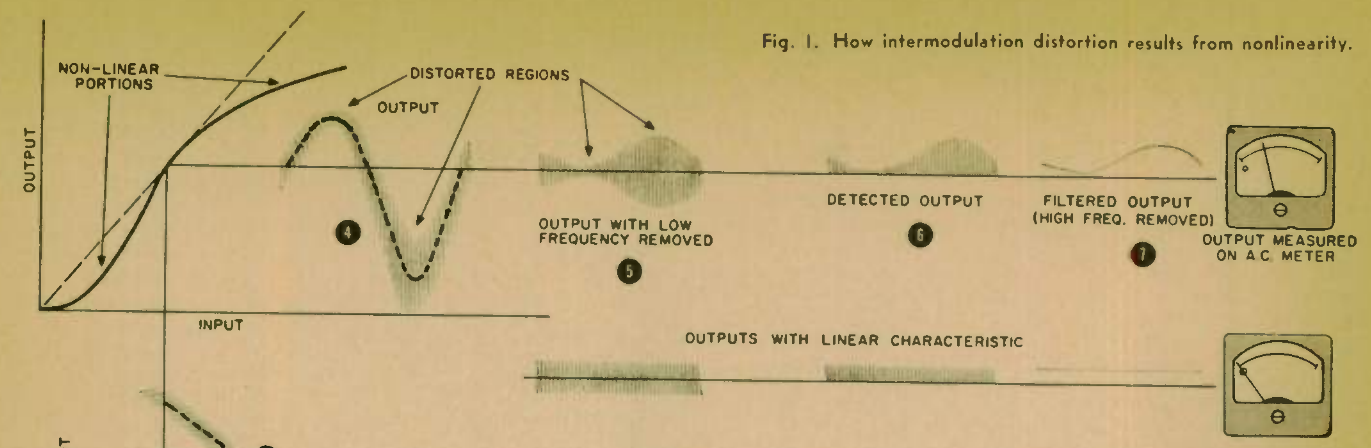
It might be of interest to point out that all of the impedance and phase shift curves shown in Figs. 3 and 4 were made using the same speaker, a twelve-inch, wide-range unit of standard make and all were made with the speaker in the same location in the room. In this manner, variables not directly concerned with the enclosure were eliminated. The primary purpose was to help in a study of the design of bass-reflex enclosures.

Interpretation of the test results has not been discussed in any detail, because that has been better covered in other recently published material both from the theoretical and design view points [1, 5, 6, 7, 8, 9] and from the more subjective viewpoint of the would-be listener [2, 3, 4].

To sum up, this simplified loudspeaker testing technique can be used to measure speaker impedance and phase shift and to give a qualitative indication of distortion generated in the speaker or its enclosure and the useful power-handling capacity of the speaker. It must also be understood that there are several things this technique will not do. One is to measure the frequency response or range of a loudspeaker (for this, one needs the more sophisticated test methods referred to at the beginning of this article). Another, and probably most important of all, it will not replace a good pair of educated ears in making the final determination as to whether the sound produced is "right."

REFERENCES

1. Augspurger, G. L.: "Loudspeaker Damping and Tonal Response," RADIO & TV NEWS, March 1957.
2. Briggs, G. A.: "All About Audio and Hi-Fi," RADIO & TV NEWS, June, July, August, September 1957, February, March 1958.
3. Crowhurst, N. H.: "Why Do Loudspeakers Sound Different?" RADIO & TV NEWS, October 1956.
4. Hartley, H. A.: "Realistic High Fidelity," RADIO & TV NEWS, May, June, July, October 1956.
5. Jordan, Edward E.: "Baffles Unbaffled," Audio, August, September 1956.
6. Moir, James: "Ported Speaker Cabinets," Audio, October 1956.
7. Sanford, Robert C.: "Bass Reflex, Design or Accident," RADIO & TV NEWS, July 1956.
8. Villchur, Edgar: "Speaker Enclosures," Audio, June, August 1956.
9. Villchur, Edgar: "Speaker Damping," Audio, October 1957.



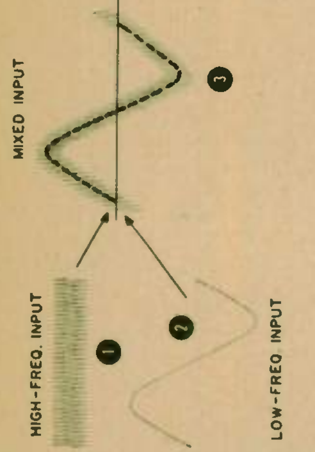


Fig. 1. How intermodulation distortion results from nonlinearity.

Intermodulation Distortion

What it is and how it's measured

By MILTON S. SNITZER / Technical Editor, ELECTRONICS WORLD

An important specification in evaluating hi-fi gear that should be thoroughly understood by the audiophile.

THE best method of checking distortion in hi-fi gear is by the use of a single input frequency in a harmonic distortion check. This method is easy to use and readily duplicated. It may not tell very much about how an amplifier would handle a somewhat more complex input. For example, if we were to apply a mixture of two input signals, one at a low frequency and the other at a high frequency, and we were to check how the amplifier handles these simultaneously and what interactions there are between them, then we would have a better indication of how the even more complex speech and music signals would be handled. This is exactly what we do when we measure intermodulation distortion (IM).

There is currently some disagreement on the value of an IM measurement. There are many in the field who feel that a total harmonic distortion (THD) test made over the entire frequency range adequately describes the amplifier's distortion performance. Following this reasoning, recent standards have been set up for THD measurements but *not* for IM measurement. Despite this, a good many manufacturers quote IM distortion figures so it is important to understand these figures and how they are obtained.

How IM Is Produced

When two different frequencies are applied to a perfectly linear device (one whose output varies directly in accordance with the input), then the output of the device contains only these two frequencies. On the other hand, when there is any non-linearity, one of the signals is changed or modulated by the other. (See Fig. 1.) When this modulation takes place, additional sideband frequencies are generated just as they are in the modulated stage of a radio transmitter. These additional sideband frequencies are *not* harmonically related to either of the original frequencies. For example, if a large 60-cps signal is combined with a small 6000-cps signal in a non-linear device, the additional frequencies of 5940 (6000−60) cps and 6060 (6000+60) cps are formed. What is more, harmonics of the original two frequencies combine with each other to produce still more "sum and difference" signals. None of these additional signals was present originally, so that we have now introduced distortion.

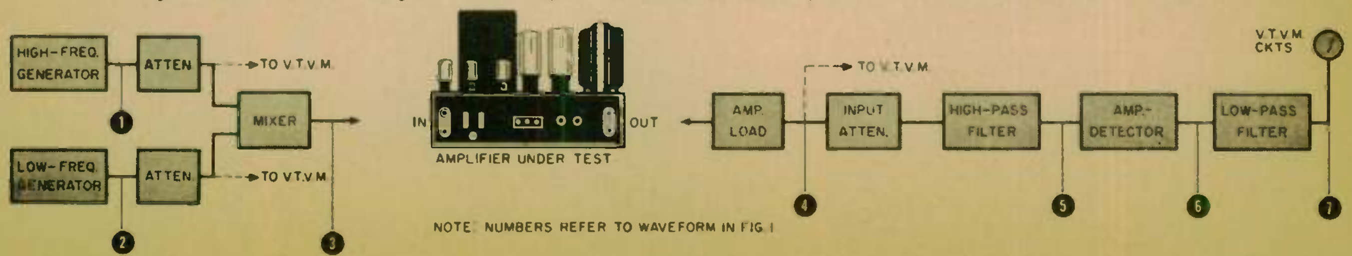
The amount of this distortion is the r.m.s. sum of all these added sideband signals expressed as a percentage of the modulated signal. In the example just given then, if the square root of the sum of the squares of all the sideband frequencies is 0.1 volt and the amplitude of the 6000-cps signal alone is 5 volts, then the IM distortion is 2 per-cent.

Audible Effects

There are two audible effects of IM distortion: one of these is the result of the amplitude modulation that occurs and the other is the result of the unharmonically related added frequencies. Assume you are listening to a Bach Chorale with the sopranos being accompanied by a loud, bass passage on the organ. If your equipment has high IM distortion, the voices may have an unpleasant, "fluttery," and unclear sound because these high frequencies are being amplitude modulated by the low-frequency tones of the organ.

In addition to this effect, any time that you listen to any combination of tones, or even a single tone that is anything but a simple sine wave, the added frequencies due to IM distortion introduce a roughness or harshness that seems to "muddy up" the original tones. Any added frequencies, whether harmonically related or not, consti-

Fig. 2. Functional block diagram of IM analyzer. Waveforms at various points in the instrument are shown in Fig. 1.



tute distortion. But when these frequencies are not harmonically related, then the audible effects are quite unpleasant.

Limits of IM

In measuring IM distortion it is common to apply to the equipment under test a large low-frequency signal along with a small high-frequency signal. The usual ratio between the two amplitudes is 4 to 1 (12 db). Low frequencies in common use are 40 cps, 60 cps, 100 cps, and 150 cps; high frequencies in common use are 3000 cps, 4000 cps, 6000 cps, and 15,000 cps.[1] Depending on which frequencies are used, and their related amplitudes, different IM distortion figures will be obtained. Therefore, it is important to state just what frequencies and amplitudes are used in a given test when specs are quoted. It is also important to state the power level at which the measurement is made.

The amount of IM distortion that should be present in top-quality hi-fi equipment has been the subject of much discussion. According to one authority[2] the following arbitrary grouping is given:

AMPLIFIER	IM
Very high-fidelity (lower test f.eq., 40 cps)	Less than 2%
High-fidelity (lower test freq., 40 cps)	Less than 4%
Good fidelity (lower test freq., 60 cps)	Less than 8%
Fairly good fidelity (lower test freq., 60 cps)	Less than 20%

Recently the *Heath Co.* has set up its own standards for IM distortion limits on power amplifiers. Its groupings, using test frequencies of 60 and 6000 cps at a 4:1 ratio, are as follows:

AMPLIFIER	IM
Professional	1% or less
High-fidelity	2% or less
Utility	3% or less

In our own lab-tested reports as well as in the specifications quoted by a large number of companies in the field, an IM distortion figure of 1 per-cent is used. If we find that an amplifier will not produce its full rated output power at 1 per-cent IM, then we down-rate the unit. Admittedly this is an extremely strict specification to meet but we feel that it is fair provided all units are subjected to the same criterion.

IM Compared with THD

In general, when an amplifier has low intermodulation distortion, it also has low total harmonic distortion (THD). When the IM is high, so is the THD. However, it must be remembered that these two methods of measuring distortion are quite different, so it is logical to expect that the percentage figures for IM and THD will not be the same.

The relation between IM and THD in a hypothetical amplifier is shown in Fig. 3. These curves are fairly typical. Note that at low power outputs (up to about 11 watts in this case) the IM distortion is less than the THD, as measured at the extremes of the frequency range to be covered (say 20 or 20,000 cps). At 10 watts output, the IM of this particular amplifier is 1% while the THD is 2%. It is quite accidental that these two figures, which we use as our own standards, occur at the same power level. But observe what happens at the higher power levels. First, the IM goes upward a little more suddenly than does THD. For this reason, many feel that the IM measurement is a much more sensitive indication of amplifier overload. Second, the two distortion curves cross each other so that the value of the IM becomes higher than the value of the THD.

If the harmonic distortion measurements are made at the mid-frequencies (400 or 1000 cps), our amplifier is seen to produce much less THD. Under these conditions, it is not uncommon for the value of THD to be less than the value of IM at all power levels. In the case of our hypothetical amplifier, an output power of 13 watts is produced at a THD of 2 per-cent. At this power level, the IM is close to 8 per-cent, or almost 4 times as great.

It is quite possible to have a totally different relation between IM and THD than is shown in Fig. 3. The design of the amplifier being checked, and particularly how it overloads, may change this picture markedly.

IM Analyzers

Although IM distortion can be measured with separate audio generators, filter circuits, oscilloscope or meter indicators, common practice is to combine many or even all the functions needed into a single instrument. This instrument may be called an IM analyzer. A block diagram of all the functions performed by such an instrument is shown in Fig. 2.

Two stable audio generators are used, one of these produces the low frequency and the other produces the high frequency required for the test. In some instruments these frequencies are adjustable, in others they are fixed. The two signals are then fed into attenuators for isolation and for adjustment of the amplitude ratio (usually 4 to 1). Some units permit the user to

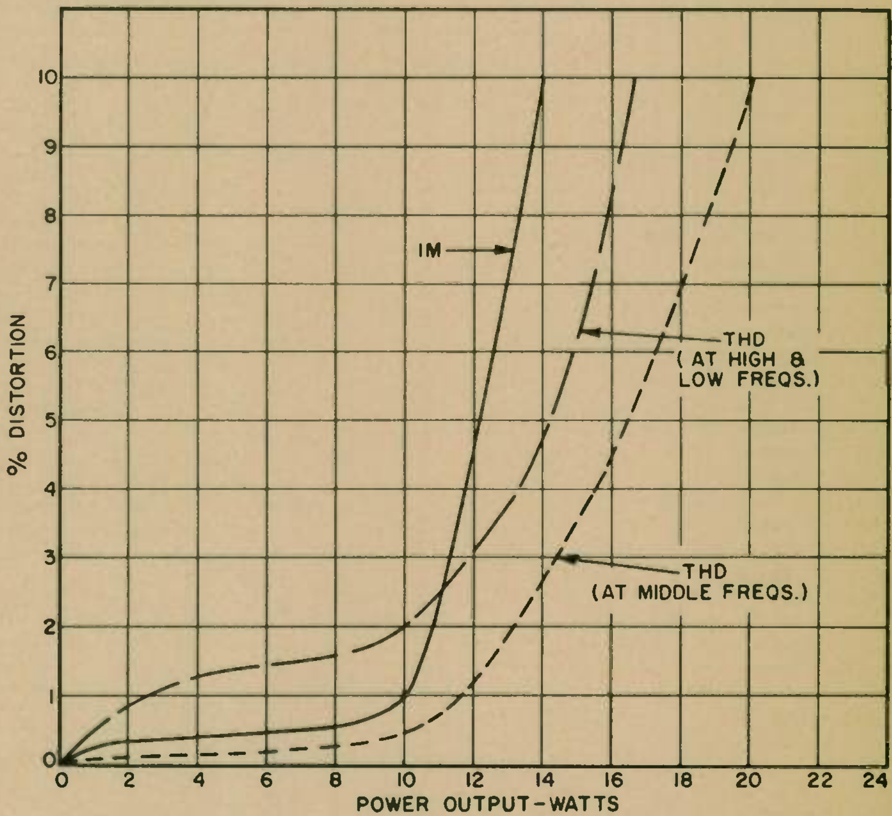
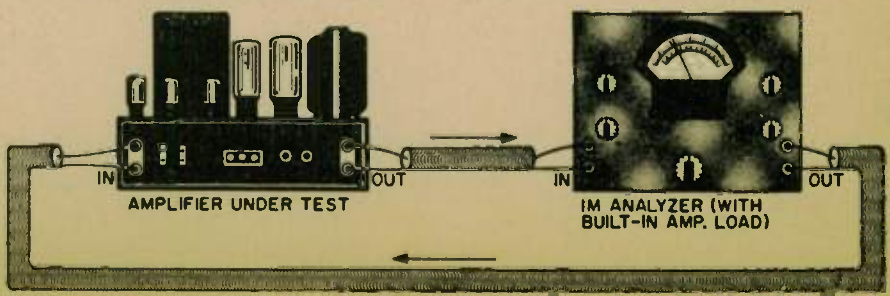


Fig. 3. Relation between IM and total harmonic distortion in hypothetical amplifier.

Fig. 4. Test setup employed for checking IM distortion of a high-fidelity amplifier.



[1] This method is the one specified by the Society of Motion Picture & Television Engineers.
[2] "Radiotron Designer's Handbook" Fourth Edition—edited by Langford-Smith.

switch internal v.t.v.m. circuits in at these points to measure the relative amplitudes of the two audio signals. After suitable mixing, the combined signal is applied to the input of the amplifier to be tested.

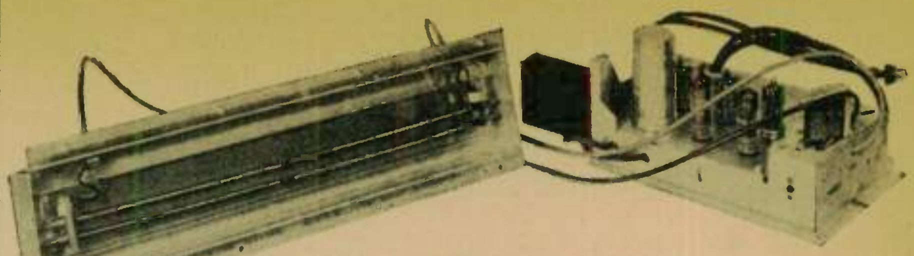
The amplifier is terminated in an output load resistor. Some IM units even have such loads built in for limited power use. The v.t.v.m. is then switched across this load so that the output power may be monitored. It is common to operate the amplifier "wide open," and simply keep increasing the mixed input signal until rated power output (or whatever power output level is desired) is obtained. See Fig. 4.

In this regard, it is important to use the same peak output voltage as would be obtained under single-frequency conditions, since it is usually at the voltage peak that distortion is generated. If the amplifier is delivering 4 volts of low frequency and 1 volt of high frequency at the same time, its total output voltage would be 5 volts. The distortion would be equivalent to that produced by a 5-volt signal. The power that should be produced by such a signal is E^2 (5^2, or 25) divided by the load resistance. On the other hand, the true power (as indicated by a voltage measurement taken across the load) is only 4^2 plus 1^2 (16 +1, or 17) divided by the load resistance. This means that the indicated power output must be multiplied by 25/17 (or 1.47) when testing for IM under these conditions to obtain an *equivalent sine-wave output power*. Hence, if the voltage measures 8 volts across an 8-ohm load, the true power is 8 watts, but the equivalent sine-wave output power is 1.47 × 8, or 11.8 watts.

The amplifier load signal is then applied through an input attenuator to a high-pass filter. This circuit removes the large-amplitude, low-frequency signal and passes on the high frequency. If the amplifier under test has IM distortion, the amplitude of this high-frequency signal will vary, perhaps as shown at the upper waveform No. 5 in Fig. 1. If the amplifier has no IM distortion, the amplitude will be constant, as shown in the lower waveform.

The high-frequency signal is then amplified and detected. (See waveform No. 6.) After detection, the signal is fed to a low-pass filter which removes the high-frequency ripple. Now the amplitude variation shows up clearly as a low-frequency signal (upper waveform No. 7). Had there been no IM distortion, the output here would be pure d.c. The remaining low frequency, representing the distortion, is then applied to the a.c. v.t.v.m. circuits for measurement. Since this meter had previously been adjusted for full-scale reading with the undistorted high-frequency applied, it will now read directly the percentage value of the intermodulation distortion.

IM distortion measurements, taken in accordance with any standard method such as the one described, allow an important specification to be quoted for evaluating high-fidelity equipment.



Two spring-like delay lines (left) and reverb amplifier produce the reverberation effect.

"Reverbaphonic Sound" System

Electro-mechanical reverberation unit to be included in Philco and Zenith packaged phonograph consoles.

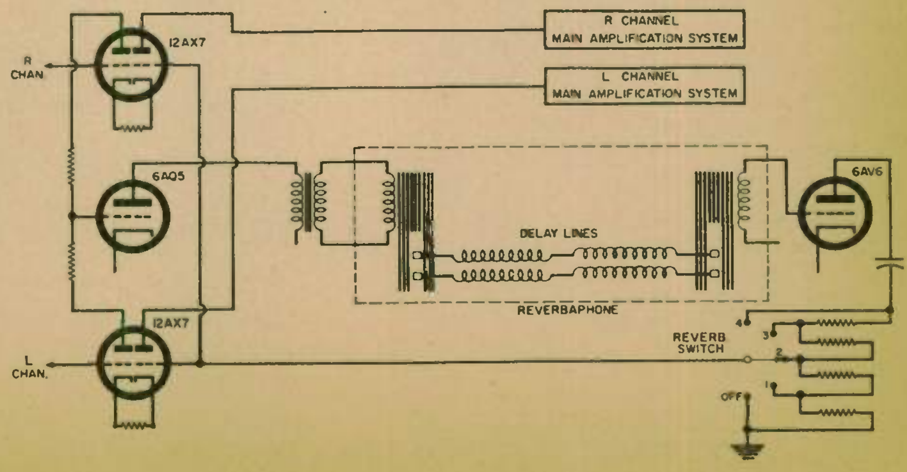
IN an effort to insert some "concert-hall reverberation" into stereo or mono sound that is reproduced in the home, two manufacturers of packaged phonographs are including, in some of their new models, an electro-mechanical reverberation unit similar in principle to the reverb unit used in *Hammond* electronic organs. Both *Philco* and *Zenith* have announced the use of the *Hammond*-designed reverberation unit in some of their new console models under the name of "Reverbaphonic Sound," for *Philco* units, or "Reverba-Tone," for *Zenith* units.

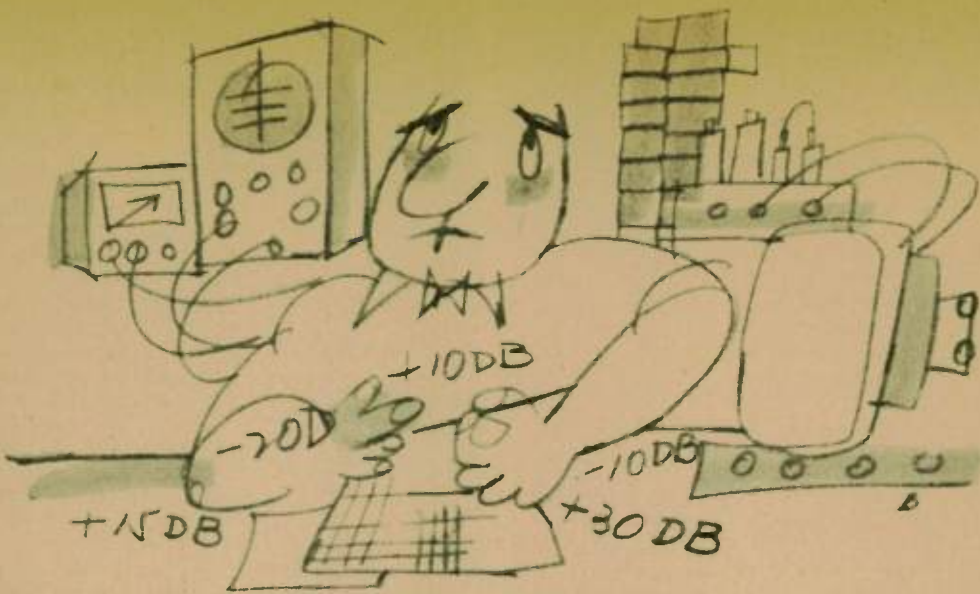
We recently attended a press demonstration of the *Philco* system and were quite impressed at the great difference in reproduced sound when the reverberation unit was switched in. A very definite and noticeable reverberation effect was heard on both stereo and mono signal sources. There may be those who argue that this is really a "gimmick" that compromises the true stereo effect, but it must be admitted that it does make for a very striking demonstration. The phonograph employed for the demonstration had its speakers for both stereo channels housed in the console. In addition, two outboard 4-inch speakers were placed in the corners of the listening room some distance away from the console. If desired, the outboard speakers, which handle signals from 200 to 300 cps and upward, may be placed in the main console. A five-position switch on the phonograph provides four degrees of reverb along with an "off" position.

Close examination of the reverberation unit disclosed what appeared to be two springs somewhat over a foot long and perhaps 3/16 inch in diameter stretched loosely between two supports (see photo above). The springs were anchored at both ends onto a pair of ferrite armatures, around which were wound a pair of coils. One of these is used for the input, and the other is used for the output.

The block diagram of the *Philco* system, shown below, indicates just where the reverberation unit fits into the over-all audio system. The time delay produced by the two springs is on the order of 30 to 40 milliseconds. The two springs have slightly different delays in order to obtain a relatively smooth delay characteristic. Because the lines are not terminated, echo signals continue to bounce back and forth for a period of up to approximately 2 seconds.

The combined left and right signals are applied through a coupling transformer to a coil wound on a ferrite armature. This armature then moves mechanically in accordance with the audio signal. The mechanical motion is transmitted along the length of the two spring-like delay lines, causing the armature at the other end of the springs to be vibrated. This, in turn, produces a signal at the output winding that may then be recombined with the original left- and right-channel signals.





The Decibel Without Pain

By **BOB ELDRIDGE**

Practical hints on understanding and using these strange units, by someone who actually likes them.

BY NOW, we are all supposed to know that the decibel is a *ratio* between two quantities of power (or voltage, or current), not a quantity of power, voltage, or current itself. Thus we should not express any known electrical quantities as being simply "so many db"—unless we are in the advertising business. The copywriter may state, "Sudso is 20 per-cent faster" and get away with it before anyone asks, "Faster than what?" However, if a technician should state, "The power at this point is 30 db," someone will invariably want to know, "30 db above or below what?"

In other words, there must be a reference level.

Sometimes this level may be understood. In the telephone industry, for example, this reference is the input level to the switchboard from a subscriber line. Other levels throughout the system, expressed simply in db, are referred to this input, with the telephone engineer primarily concerned with the magnitude of the losses in the various portions of the path between the microphone of one telephone and the receiver of another.

He is also interested in the gain in db of each amplifier in the chain that will restore the losses: what goes down must be brought back up again if Mom in New York is to hear Johnny's plea from camp for just another few dollars. In the detailed engineering of the circuit, he also becomes interested in the actual rather than relative power present at various points in the path, but that is a matter for db's twin brother, dbm, of whom more will be heard later.

For the moment, let's consider only that the db is a ratio. An amplifier that produces 100 watts of output when 1 watt is fed into it is a 20-db amplifier. But so is another that produces 100 milliwatts out from 1 mw. in. From this comparison, we see that a 20-db amplifier is one that has a power gain of 100 times without regard to its actual power-handling capacity.

What About Logs?

The key to the whole db business, as many technicians unhappily acknowledge, is a working familiarity with logarithms. A great deal can be done in handling decibels without using logarithms, as will be noted later. However, those practical men who have been getting along without logs for years just don't realize what they are missing. Logs are very easy to use—they make multiplication and division less of a chore. In fact, logs literally *turn* multiplication *into* addition, and division into subtraction. This same great advantage is present in the use of db. Although a full discussion of the use of logs in general calculations will be avoided, it will be useful to review some facts concerning the logarithm to the base 10 before getting to the meat of the dog biscuits (db).

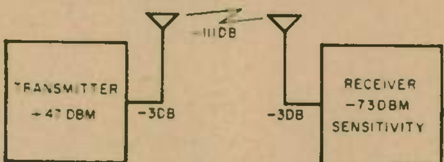
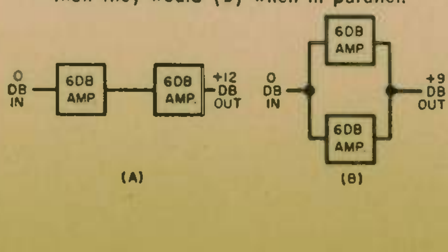


Fig. 1. Gains and losses in a transmission link are represented as shown to determine adequacy of signal level.

Fig. 2. Two identical amplifiers connected (A) in series yield greater gain than they would (B) when in parallel.



(A) (B)

The logarithm of a number is the power to which 10 must be raised to obtain that number. 100 is 10 squared (10^2), or 10 to the power of 2. Therefore, the power to which 10 must be raised to make 100 is 2; so the log of 100 is 2. To take another example, 2 is 10 to the power of .3 (to be precise, .3010, but who wants to be precise at a time like this?). To find this value, or to convert a number to a log or the reverse, either a table of logarithms or a slide rule may be used.

To multiply several numbers together, we merely find their logs, add these together, and translate the new log thus obtained back to a number again. Thus, using the two logs we have already examined, let us multiply 100x100x100x2x2. With logarithms, this becomes 2+2+2+.3+.3, which quickly adds up to 6.6, the new logarithm. An antilog table or a slide rule tells us that the number of which 6.6 is the logarithm is 4×10^6, or 4,000,000.

Now for tying up logarithms with decibels—in formula: $db = 10 \log P_1/P_2$.

"Which means?" as the straight man might say to his partner.

"A decibel is ten times the log of the ratio between two power levels."

"Why ten times?"

"Because the bel, the first unit for measuring the relative level between two sounds, was too large to handle; so they chopped it into ten parts and called each a decibel ($\frac{1}{10}$ bel)."

From the formula it is obvious that the db is logarithmic. In this fact lies many great advantages. Power ratios of millions to one can be expressed in simple numbers and vast products can be obtained by merely adding simple numbers together. If you think that this is rarely necessary, consider that a difference of only 60 db represents a power ratio of a million to one.

Table 1 shows some of the easily remembered and frequently used power ratios, expressed as numerical ratios and in equivalent db. With these figures, many complicated calculations can be reduced to mental arithmetic. For example, whenever a given power level is multiplied by 10, we add 10 db (or, when it is divided by 10, subtract 10 db). To double any power, add 3 db. Since quadrupling a power is doubling it twice, this comes to 6 db (3+3). Just by manipulating this table, many other ratios can be deduced. For example, to increase a power ratio 20 times, we must increase it 10 times (10 db) and then double it (3 db). Calculation will show that a power increase of 20 times is indeed a change of 10+3, or of 13 db.

Other examples: To obtain 5 times a power, we can find 10 times that power and then halve it. In db, this would be +10−3, or +7 db. To increase by 50 times, we could increase by 10 times (+10 db) and then by 5 times (+7 db). The answer would be +17 db. To find 25 times (half of 50 times), we know that +17−3=+14 db.

The Dbm

The standard test tone used in the telephone industry is a 1000-cps signal with a power of 1 milliwatt. Since this is a convenient reference level, the term dbm, meaning "db with reference to 1 milliwatt," is now in widespread use. Since there is a specific reference point (1 milliwatt is always zero dbm), any figure given in dbm is *not* a ratio. It is a definite quantity of power. Thus 1 watt (1000 mw.) can be written as +30 dbm; and 1 microwatt is −30 dbm.

The dbm is very useful in system calculations. For example, let us take an imaginary u.h.f. link that must be set up. We have the following: 1. a 50-watt transmitter, 2. a receiver that requires an input of 50 microvolts of signal for efficient operation, 3. antenna feeders and connectors that entail a loss of 3 db at each end, 4. a required safety margin of 20 db to allow for fading, and 5. attenuation of the signal in the air path between transmitter and receiver shown to be 111 db. We draw a block diagram, as in Fig. 1.

A 50-microvolt signal across the 50-ohm input impedance of our receiver tells us we need a power here of −73 dbm. This is another way of stating the receiver's sensitivity. The transmitter output, already given in watts, can be restated as +47 dbm. If we forget everything else (losses) in between transmitter and receiver, the *ratio* between the transmitter output and the power required at the receiver input is 120 db (not dbm). This is the *difference* between transmitter output and receiver input, which is +47−(−73).

This 120 db is called the system gain, and represents the maximum amount of attenuation that could be tolerated

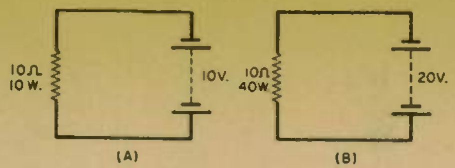


Fig. 3. Doubling voltage (B) applied across a circuit (A) also doubles current. Thus the power drop quadruples.

between transmitter and receiver without degrading performance. Now, on the debit side, we have: 3-db loss in the transmitter antenna feeder system, 111-db loss in the air path between these stations, and 3-db loss in the receiver antenna feeder system. These losses add up to −117 db.

Thus, if the antennas used show unity gain (such as conventional dipoles), the system will work with 3 db to spare. However, we are looking for a 20-db safety margin to allow for fading. We must therefore use antennas that will give us a total gain of 17 db. Antennas at each end with a gain of 8.5 db each will do the trick.

Now here is a point worth thinking about: If we left the unity-gain antenna at the transmitter and stacked two 8.5-db antennas at the receiver without losses, we would fall short of the gain requirement. We would get, not 17 db, but only 11.5 db! Let's consider this statement in a more familiar form—the case of the two amplifiers in Fig. 2. In series, with one amplifier *multiplying* the output of the other, the logarithmic increase is such that, expressed in db, we get twice the gain of one amplifier. In parallel (Fig. 2B), we only double the power. This (Table 1) is a gain of only 3 db over the use of one amplifier or antenna. In the case of the amplifiers, this could easily be measured.

To return to the dbm—it is so useful precisely because it indicates a definite power and yet can be manipulated in a logarithmic manner along with ordinary decibels. Transmitters, amplifiers, receivers, microphones, and loudspeakers can all be rated, as to sensitivity or output, in dbm, even though the impedance of each type of unit may be different. Other units similar to the dbm include dbw (referred to a level of 1 watt), dbk (in reference to a kilowatt), and dbx, a unit of crosstalk between two circuits. There are also others.

Voltage, Current Ratios

So far, we have talked about the ratio between two powers. The ratio between two voltages, or between two currents, can also be expressed in db, where these are the quantities that are known—but the formula is different. This stems from the fact that, when you relate a given power to the voltage or current that is involved with it, the relationship is not linear. There is always a square involved (power, in watts, is I^2R or E^2/R). Again, we look at the matter in practical terms, with the help of Fig. 3.

If we put a 10-ohm resistor across a 10-volt supply, 10 watts of power (Fig. 3A) will be dissipated in the resistor. If the voltage is doubled (Fig. 3B), the power in the resistor is not doubled—it is quadrupled. This is because, following the rule made public by Mr. Ohm, the current is doubled as well as the voltage.

This brings us to another principle in the use of logarithms: to square any number, multiply its log by 2, then find the antilog of this product. Since a square is involved when we go from a power to a voltage or a current, we go back to our old db formula (Ugh! There he goes again!) for power ratios and multiply the logarithm by 2. Thus, for converting two voltages to a db ratio, the formula is $db = 20 \log E_1/E_2$. However, a db is a db. If you compare two specific signals, the respective powers will be different from the respective voltages, but the db ratio will always be the same.

The point may be further illustrated by reference to Table 2, where the db equivalents for many voltage ratios are given. If the voltage across a given load is doubled, there is an increase of 6 db. Note that doubling the voltage across the load will have quadrupled the power. Thus a 4-time increase in power (Table 1) is a gain of 6 db.

Reading a Db Meter

A word of caution is necessary when reading db directly on a meter. The latter is actually a voltmeter, not a power meter, and is therefore calibrated for readings across a specific impedance. For relative readings, this is not much of a problem. For instance, if one should start out with a reading of +5 db across any circuit, and this reading should change to +3 db during tests across the same impedance, the change will always be a difference of 2 db.

For reading dbm however, the impedance for which the meter was calibrated must be known. In most cases, this is "0 db—1 mw. in 600 ohms," usually printed on the meter face. However some older meters are calibrated to make zero db equal to 6 mw. in 500 ohms. In either case, this direct reading is obtained on one range,

POWER RATIO	DB	POWER RATIO	DB
One thousandth	−30	One eighth	−9
One hundredth	−20	One quarter	−6
One tenth	−10	One half	−3
Unity	0	Unity	0
Ten times	+10	Doubled	+3
One hundred times	+20	Quadrupled	+6
One thousand times	+30	Eight times	+9

Table 1. Common power ratios in db.

Table 2. Common voltage ratios in db.

VOLT. RATIO	DB	VOLT. RATIO	DB
One hundredth	−40	One fourth	−12
One tenth	−20	One half	−6
Unity	0	Unity	0
Ten times	+20	Doubled	+6
One hundred times	+40	Quadrupled	+12

and direct readings on other ranges are corrected by adding or subtracting so many db, as specified by the manufacturer, where separate db scales are not used.

If the impedance at which the db scales have been calibrated is in doubt, a check can be made against the voltage scales. For example, if .775 volt corresponds to zero db, the meter has been calibrated to the modern standard, and powers in a 600-ohm circuit can be read directly off the meter in dbm. Such readings can be made across other impedances too, simply by applying a correction factor. For example, let us suppose the meter is calibrated for 600 ohms. In reading across a 1200-ohm circuit, one would simply subtract 3 db from the scale reading to obtain the power in dbm. Across a 300-ohm circuit, add 3 db to the reading; across 150 ohms, add 6 db; and across 75 ohms, add 9 db. Correction factors can be worked out for any impedance.

The important thing is to start using db wherever possible, even when they are not needed. The familiarity thus acquired will simplify matters when db becomes essential.

TRANSISTOR AMP HINT

By CHARLES ERWIN COHN

IT IS considered good practice not to operate a transistor audio power amplifier with its speaker disconnected. The reason for this is that, with the speaker disconnected, signals can build up considerable voltage across the unloaded output transformer. Such peaks might exceed the collector voltage rating and burn up the transistor.

Therefore, when external speakers are used with such amplifiers, as in car radios for example, it is a good idea to make the speaker connection through a phone plug and shorting jack for very good reason.

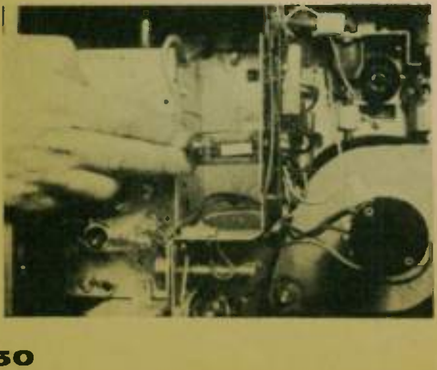
This will short the secondary of the output transformer when the speaker is removed and thus prevent a build-up of dangerous voltages.

TUBE HELP

By ROBERT HERTZBERG

A SMALL strip of tape of any kind, positioned on a miniature tube to indicate the indexing space between pins 1 and 7 or 1 and 9, is of great help in getting the tube into its socket without mashing the pins. Leave it there; it might be useful again some other time.

Masking tape, friction tape, or even the end of a Band-Aid serve the purpose nicely. —30—



Measuring Your Audio Power Output

By HAROLD REED

Comparative tests with load resistor and voltmeter and voltmeter-ammeter method show surprising results.

WHEN checking the power output of an audio amplifier, it is customary to use a substitute load resistor of ohmic value equivalent to the amplifier output impedance. The temperature of the load resistor, unless of considerably higher power rating than the power output of the amplifier under test, will rise rapidly and its rated resistance value may change enough to result in erroneous measurements. This deviation can be serious when making critical distortion tests.

Suppose, as an example, an amplifier rated at 20 watts output is being checked with a 4-ohm resistor load. The amplifier output would be adjusted to obtain 8.95 volts across this load. Thus, it would be said that $W = E^2/R$, or $8.95^2/4 = 20$ watts. Also, suppose the load value rises due to heating by just 0.7 of an ohm. The author has seen this occur when using a good sized resistor for the load. Then, with this same output voltage, the power would be $8.95^2/4.7$, or only 17 watts. A distortion measurement made under these conditions may look very good and it would be assumed that it was taken at the 20-watt level. This same amplifier may severely distort the signal waveform as the maximum power output is approached.

Therefore, to check a 20-watt amplifier at rated output, make certain it is delivering 20 watts to the load. Sometimes, the last few watts can change the scope waveform pattern from a Jekyll to a Hyde. Heating of the load resistor must be avoided. Its power rating should, preferably, be at least 5 times the power output of the amplifier.

Amplifier power output measurements have been made by the voltmeter-ammeter method. The voltmeter is connected across the load resistor as before and the ammeter is inserted in series with this load. Then, it would be assumed that $W = EI$. This can prove to be even more erroneous.

Ordinary a.c. ammeters are designed for low-frequency applications and although their accuracy may be given as 2% of full scale, the deviation from this tolerance value is considerable with increasing signal frequency.

An example of this discrepancy is shown in Table 1. The data given in this table was obtained from measurements made with a representative low-frequency a.c. meter, 0-2 amperes. A load resistor of known ohmic value and high power rating, to avoid heating, was used in these measurements. It is readily seen that $W = EI$ does not equal E^2/R except in one case. This equality is observed only at a frequency of 30 cycles.

Using a low frequency as described, both methods may be employed to provide a double check on the power output figure. When the amplifier is first turned on, note the E^2/R and $E \times I$ values which should result in about the same power output figure. After running the amplifier for about 15 minutes re-adjust the output voltage, if necessary, to the level first noted. Again note the product of $E \times I$ and, if it is not equal to E^2/R, then the load resistor has changed value due to heating.

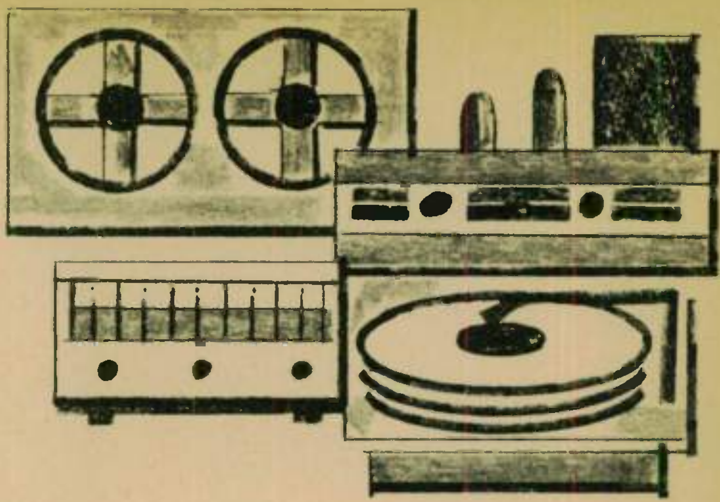
As mentioned previously, most a.c. meters have a full-scale accuracy within 2%. In the previous example given for a 20-watt amplifier, if the meter should be off by the full 2% either on the high side or low side, the indicated power output would be 20.4 watts or 19.6 watts, respectively, a negligible deviation from true power output. It is advisable to work in the upper half of the meter scale to minimize error.

Some manufacturers offer, for an additional charge, a.c. meters designed for operation at higher frequencies. *Weston*, for example, will supply meters designed for operation up to 2500 cycles.

When using ordinary type a.c. meters for audio power measurements, stick to frequencies under 100 cycles or apply a suitable correction factor which can be derived by obtaining test data on the particular meter as shown in Table 1.

Table 1. These are the results of the a.c. ammeter test performed by the author. A 4-ohm load resistor was employed and the output voltage of the amplifier was held at a constant 4-volt output. The power being delivered in this case was 4 watts.

FREQ. (cps)	OUTPUT CURRENT (I)	OUTPUT POWER (W)
30	1.0 amp.	4.0
50	.99	3.96
100	.98	3.92
500	.96	3.84
1000	.95	3.80
10,000	.80	3.20
20,000	.65	2.60



SECTION·3

Amplifiers • Tuners • Phonos • Tape

Power Output Ratings For Hi-Fi Amplifiers .. 54

Compact 20-Watt Hi-Fi Amplifier 57

Bass-Treble Hi-Fi Amplifier 60

Low Noise Transistor Preamp 65

Tape System For Stereo 68

Transistorized FM Multiplex Adapter 71

Multiplexing Music With One Recorder 74

New AM-FM Tuner Standard 78

Simple Booster For FM Tuners 81

A Sweep Generator For Hi-Fi AM 84

America's Most Popular, Most Authoritative Books on High Fidelity, Stereo and Tape

Here are some of the world's greatest hi-fi books . . . chosen carefully by Ziff-Davis Electronics Book Service as among the best in their field.

Right now, one or more of these great books will be sent to you for 7 days FREE! Simply write your choices on the coupon below and mail it today. When your books arrive, read and enjoy them for seven full days. If, after that, you don't agree that they are everything you need and want, return them and owe nothing.

2751. HI-FI GUIDE—STEREOPHONIC SOUND, Hoefler

A "how-to" book on hi-fi, written in simple language. Will help you buy the right equipment and see that you get the most out of your stereo or monaural investment. $2.50

2752. HIGH QUALITY SOUND REPRODUCTION, Moir

The perfect manual for both the professional engineer and the serious amateur interested in high fidelity. The "why" and "how" of sound reproduction is covered in complete detail. $15.00

2753. LOW-COST HI-FI, Hoefler

Hundreds of hints for budget hi-fi will be found in these fourteen chapters with over 300 detailed photographs, drawings and diagrams. Will save you money in starting or improving your system. $2.50

2755. THE PRACTICAL HI-FI HANDBOOK, King

A guide to high fidelity sound reproduction for the service engineer and amateur. Chapters on amplifiers, loudspeakers, pickups, microphones, record players, disc, tape and stereo. $5.95

2756. REPAIRING RECORD CHANGERS, Ecklund

A practical manual on repair of mechanical elements of record changers, including pickups, needles, changer actions, motors, drives, tripping, dropping and shut-offs. Also magnetic recorder repairs. $5.95

2760. HI-FI STEREO FOR YOUR HOME, Whitman

Tells what stereo is, how it differs from hi-fi, how it works, how it affects home listening habits, and how to install and maintain it. Complete list of terms defined. Generously illustrated. $3.50

42. REVERE TAPE RECORDER GUIDE, Tydings

The first non-technical book to provide useful information on the Revere Tape Recorder. Also a basic guide to the entire field of tape. Will show you new uses and add to your enjoyment. $1.95

49. TAPE RECORDING GUIDE, Marshall

Designed to help you get the most out of your tape recorder, whether for business, pleasure or professional use. A handy guide to have around, no matter what equipment you own. $1.95

2750. ELEMENTS OF MAGNETIC TAPE RECORDING, Haynes

Here's how to get professional results with tape the way the experts do. Complete nomenclature, basic techniques, how to splice and edit, how to repair and maintain your recording equipment. $7.95

2754. MAGNETIC TAPE RECORDING, Spratt

Designed to give principles of magnetic recording and to enumerate characteristics of both the medium and the machines. Excellent for adapting magnetic recording to special needs and wider applications. $8.95

2757. RIBBONS OF SOUND, Barleben

A handbook on the fundamentals of magnetic tape recording simply and interestingly presented. Factual information you can use no matter what type or make of recorder you own. Paper. $2.50. 2772. Cloth. $3.50

2758. TAPE RECORDERS AND TAPE RECORDING, Weiler

An ideal sourcebook of information on all aspects of tape recording. Covers all fundamentals necessary to realize full potential of your tape equipment. Special sections on accessories. $2.95

2000. STEREO HI-FI GUIDE, 1960 Ziff-Davis

1960 edition features 60-page exclusive by Joseph Marshall on components and how they work. Includes "what you should know before buying stereo". Complete, interesting, invaluable! $1.00

2002. ELECTRONIC KITS DIRECTORY, 1960, Ziff-Davis

New 1960 edition lists over 750 kits, latest models, prices and features for hi-fi kits—preamps, amplifiers, tuners, speakers — ham radio, SWL, Citizens Band. Fun and education. $1.00

2004. HI-FI ANNUAL & AUDIO HANDBOOK, Ziff-Davis

1960 edition. Prepared by the editors of Electronics World. An excellent advanced guide to theory, construction and circuitry. Over 40 pages on stereo amplifiers and equipment. $1.00

2005. HI-FI DIRECTORY & BUYERS' GUIDE, Ziff-Davis

1960 edition. World's only complete listing of all hi-fi and stereo equipment, components and accessories. Five big sections, data on more than 1200 components, over 1,000 illustrations! $1.00

2006. ELECTRONIC EXPERIMENTER'S MANUAL, Findlay

With a few dollars worth of basic tools and this book to guide you, you can explore the wonder-world of electronics experimentation more completely than ever before. 10 big sections. $4.95

2769. THE ELECTRONIC MUSICAL INSTRUMENT MANUAL, Douglas

Covers every design phase of the modern electronic musical instrument — including theory, schematics of organ circuits, the science of sound as well as the art of music. $7.50

Yours For a 7-Day Free Examination from ELECTRONICS Book Service!

Each volume is designed to help you get more use and pleasure from your high fidelity equipment. Whether you're planning to buy or ready to improve your system—whether you now enjoy stereo or plan to convert to stereo—whether you're a music-lover or a hi-fi do-it-yourselfer—you'll find one or more books of interest below! For yourself or for gift-giving—use the coupon below today!

2762. FROM MICROPHONE TO EAR, Slot
A complete survey of the technique of sound recording and reproduction, discussing the entire chain from microphone to loudspeaker. A practical book for music-lovers as well as sound technicians. $4.95

2764. ELEMENTS OF SOUND RECORDING, Frayne and Wolfe
A discussion of basic problems in sound recording and reproducing for the designer, engineer, technician and student. Covers acoustics, vacuum tubes, audio amplifiers, etc. 686 pages. 483 illustrations. $12.00

2766. HIGH FIDELITY HOME MUSIC SYSTEMS, Wellman
Authoritative advice on choosing the right type of system for your needs with practical information on the functions of each unit—selection, assembly and installation. $4.50

2767. HIGH FIDELITY TECHNIQUES, Newitt
A comprehensive treatment of both the principles and practice of modern high fidelity sound systems — how to plan a new system, how to improve an existing one. A goldmine of hi-fi information. $7.50

2771. HI-FI HANDBOOK, Kendall
How to plan your home music system, choose the best components, install your system easily and maintain it by yourself. All these, and ways to save money, are presented in this basic book. $3.50

2759. TECHNIQUES OF MAGNETIC RECORDING, Tall
Translates the complexities of a science into practical, easy-to-follow techniques. New ideas, new standards, especially for the amateur who wants a good working knowledge of magnetic recording. $8.50

2763. MAGNETIC RECORDING TECHNIQUES, Stewart
Covers the technology of magnetic recording methods and devices for engineers and technicians concerned with their application in audio, TV, communications, computers and other fields. $8.50

2765. YOUR TAPE RECORDER, Marshall
Based on 2500 experiments with almost every type of recorder, this book helps to eliminate trial and error under all conditions. Includes illustrations of 55 magnetic recorders with specifications. $4.95

2768. MAGNETIC RECORDING, Begun
Thoroughly covers the theory of magnetic recording, various types and makes of recorders, their applications and performance measurements. Includes chapter on important research problems. $5.00

2770. HOW TO USE A TAPE RECORDER, Hodgson and Bullen
Written to help business and home recorder owners to learn how to get full value from their machines. Explains basic working of tape recorders and accessories as modern tools of communication. $4.95

See Your Hi-Fi Dealer Or Use This Coupon Today!

Leading hi-fi dealers and salons and radio and electronics parts jobbers are making their stores headquarters for books on every electronics subject. You can take this list to your favorite dealer for immediate purchase.

If your local dealer does not carry books, use the coupon for prompt delivery from ELECTRONICS BOOK SERVICE, on a 7-day free trial basis.

ELECTRONICS BOOK SERVICE, One Park Avenue, New York 16, N. Y.

Please send me the book(s) I have listed below for a FREE 7-day Trial Examination. I understand that if I am not completely satisfied, I may return my selection(s) and I'll owe you nothing. Otherwise, I will send you payment for the book(s) of my choice, plus postage and handling. 1706

NUMBER	TITLE	PRICE

(If you need more space to list additional titles, attach a sheet of paper with additional list.) *TOTAL ______

☐ SAVE MONEY! Enclose payment in full for the book(s) of your choice and we will pay shipping charges. Same return privileges and prompt refund guaranteed.

☐ Please send me FREE CATALOG, when published

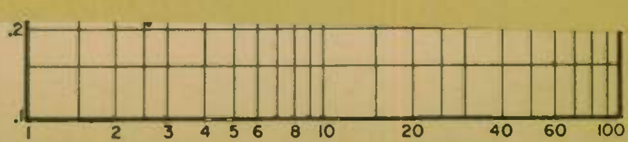
*New York City Residents, please add 3% sales tax.

NAME ______________________
PLEASE PRINT CLEARLY

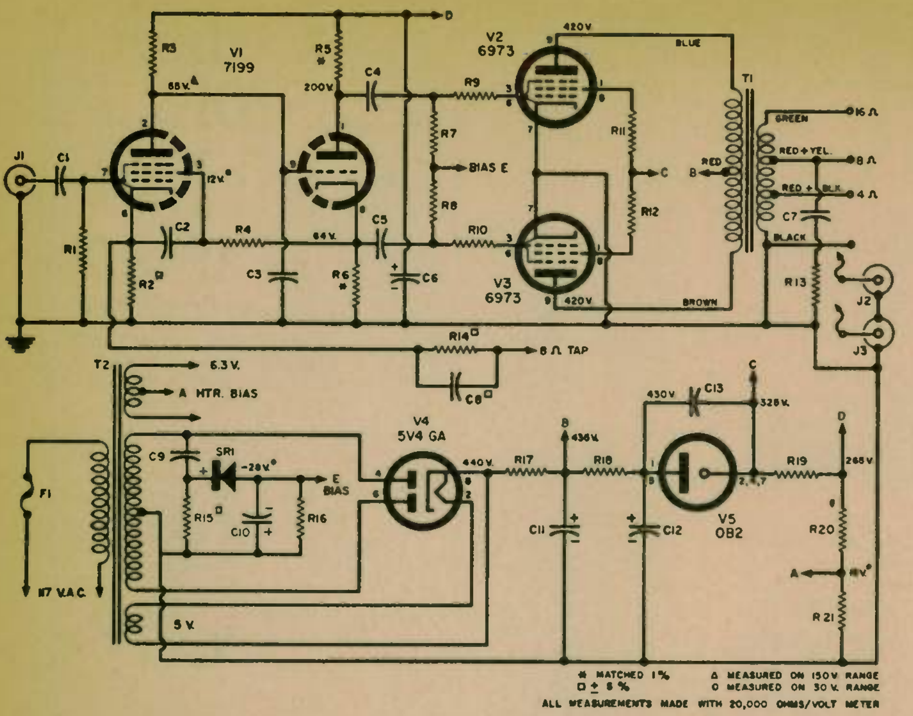
ADDRESS ______________________

CITY ______________ ZONE ______ STATE ______________

use of peak-indicating voltmeters, but it is somewhat easier to obtain peak power simply by multiplying r.m.s. power by 2. For practical purposes, "peak power" is not significant in evaluating amplifier performance. It is unfortunate that the term "peak" in



$E_{R.M.S.}$ (READ ON V.T.V.M.)



R_1—330,000 ohm, ½ w. res.
R_2, R_{13}—1000 ohm, ½ w. res.
R_3, R_4—820,000 ohm, ½ w. res.
R_5, R_6—15,000 ohm, 1 w. res. (matched 1%).
R_7, R_8—270,000 ohm, ½ w. res.
R_9, R_{10}—470 ohm, ½ w. res.
R_{11}, R_{12}—100 ohm, ½ w. res.
R_{14}—33,000 ohm, ½ w. res.
R_{15}—5600 ohm, 1 w. res.
R_{16}—56,000 ohm, ½ w. res.
R_{17}—56 ohm, 1 w. res.
R_{18}—470 ohm, 1 w. res.
R_{19}—10,000 ohm, 1 w. res.
R_{20}—150,000 ohm, 1 w. res.
R_{21}—10,000 ohm, ½ w. res.
C_1—.05 µf., 400 v. capacitor
C_2—.5 µf., 200 v. capacitor
C_3—15 µµf. mica capacitor
C_4, C_5—.1 µf., 400 v. capacitor
C_6—20 µf., 350 v. elec. capacitor
C_7, C_{13}—.02 µf., 200 v. capacitor
C_8—82 µµf. mica capacitor
C_9—.05 µf., 600 v. capacitor
C_{10}—100 µf., 50 v. elec. capacitor
C_{11}, C_{12}—40 µf., 450 v. elec. capacitor
J_1, J_2, J_3—Phono jack
SR_1—75 ma. selenium rectifier
F_1—2 amp fuse
T_1—Output trans. 8000 ohms c.t. to 16, 8, 4 ohm v.c. (Triad S-31A)
T_2—Power trans. 350-0-350 v. @ 120 ma.; 5 v. @ 3 amps; 6.3 v. c.t. @ 4.7 amps. (Thordarson 24R05U or equiv.)
V_1—7199 tube (RCA)
V_2, V_3—6973 tube (RCA)
V_4—5V4-GA tube (RCA)
V_5—0B2 tube

Fig. 4. Circuit of the miniature high-fidelity amplifier, with typical voltages.

this stage and of the direct-coupled phase-splitter stage were carefully chosen to assure class A operation of these stages at signal levels well beyond that required for full output. Grid-No. 2 voltage for the pentode unit is obtained from the cathode of the triode unit. Because of the large time-constant of the grid No. 2 circuit (0.82 megohm x 0.5 µf.), the grid-No. 2-to-cathode potential is essentially constant for audio frequencies. This arrangement provides feedback between the two units which stabilizes the operating point of the phase-splitter against changes in the "B" supply voltage and tube characteristics.

Two networks are used to assure stability at high frequencies. The 15-µµf. capacitor connected from the plate of the pentode amplifier to ground provides a gradual roll-off in response beginning at 15 kc. The .02-µf. capacitor and the 1000-ohm resistor across the 8-ohm winding of the output transformer suppress high-frequency oscillations under overload conditions.

Power-Supply Details

The excellent performance of the amplifier is partly due to the design of the power-supply circuit. An *RCA* 5V4-GA rectifier tube is used because of its small internal voltage drop and excellent regulation. Its indirectly heated cathode also delays the build up of full "B" supply voltage until the tubes have reached operating temperature, thus minimizing voltage surges across the power-supply components. Grid-No. 2 voltage for the 6973 output tube is obtained from the plate-supply voltage through an 0B2 glow-discharge voltage-regulator tube. This arrangement provides the extremely stable grid-No. 2 voltage necessary to assure low distortion at full output and minimizes the current drain on the power transformer because it eliminates the need for power-wasting, voltage-dropping resistors or bleeder networks.

Construction

Fig. 1 is a top view of the amplifier while Figs. 2 and 3 show the underside and wiring. With the exception of the relative positions of the power and output transformers, placement of components is not critical.

No marked effect on the amplifier's stability or performance was noted despite the fact that it was built in the rather limited confines of the 4" x 5" x 6" "Minibox." The windings of the output transformer are well shielded electrostatically by the core and the transformer housing. Coupling from the magnetic field of the output transformer is not a problem because this field is of very small magnitude. To minimize undesired coupling from the plate lead of the output transformer, the transformer was mounted so that these leads come out of the case on the side farthest from the input stage.

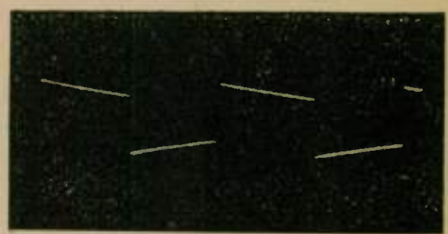
Because of its proximity to the power transformer, the output transformer picks up a small a.c. component. The resulting hum has a level of about 100 db below 20 watts and can be reduced, if desired, by rotating the output transformer 5 to 10 degrees while observing the a.c. output of the amplifier on an oscilloscope or a high-sensitivity voltmeter.

Compact construction often engenders heat-dissipation problems. In this amplifier the principal sources of heat, *i.e.*, the tubes and power transformer, are mounted on top of the chassis so that the heat rises without detrimental effect on the components on the underside of the chassis.

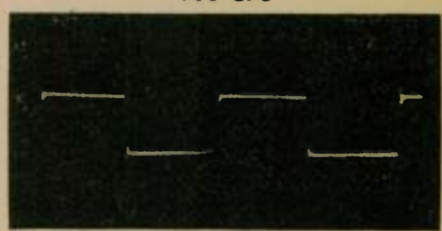
To increase the flexibility of this amplifier, two phono jacks are mounted on the rear of the chassis and connected to two spare terminals on the output-terminal strip. Any tap on the voice-coil winding can thus be connected to either jack by connecting a jumper between the appropriate binding posts.

The following suggestions will help

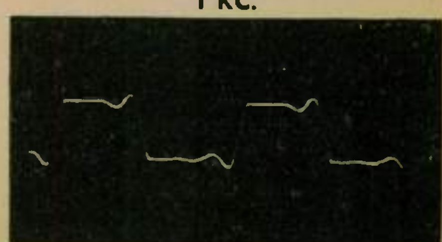
Fig. 5. Square-wave response of amplifier.



100 CPS



1 KC.



10 KC.

Yours For a 7-Day Free Examination from ELECTRONICS Book Service!

Each volume is designed to help you get more use and pleasure from your high fidelity equipment. Whether you're planning to buy or ready to improve your system—whether you now enjoy stereo or plan to convert to stereo—whether you're a music-lover or a hi-fi do-it-yourselfer—you'll find one or more books of interest below! For yourself or for gift-giving—use the coupon below today!

2762. FROM MICROPHONE TO EAR, Slot

A complete survey of the technique of sound recording and reproduction, discussing the entire chain from microphone to loudspeaker. A practical book for music-lovers as well as sound technicians. $4.95

2764. ELEMENTS OF SOUND RECORDING, Frayne and Wolfe

A discussion of basic problems in sound recording and reproducing for the designer, engineer, technician and student. Covers acoustics, vacuum tubes, audio amplifiers, etc. 686 pages. 483 illustrations. $12.00

2766. HIGH FIDELITY HOME MUSIC SYSTEMS, Wellman

Authoritative advice on choosing the right type of system for your needs with practical information on the functions of each unit—selection, assembly and installation. $4.50

2767. HIGH FIDELITY TECHNIQUES, Newitt

A comprehensive treatment of both the principles and practice of modern high fidelity sound systems—how to plan a new system, how to improve an existing one. A goldmine of hi-fi information. $7.50

2771. HI-FI HANDBOOK, Kendall

How to plan your home music system, choose the best components, install your system easily and maintain it by yourself. All these, and ways to save money, are presented in this basic book. $3.50

2759. TECHNIQUES OF MAGNETIC RECORDING, Tall

Translates the complexities of a science into practical, easy-to-follow techniques. New ideas, new standards, especially for the amateur who wants a good working knowledge of magnetic recording. $8.50

2763. MAGNETIC RECORDING TECHNIQUES, Stewart

Covers the technology of magnetic recording methods and devices for engineers and technicians concerned with their application in audio, TV, communications, computers and other fields. $8.50

2765. YOUR TAPE RECORDER, Marshall

Based on 2500 experiments with almost every type of recorder, this book helps to eliminate trial and error under all conditions. Includes illustrations of 55 magnetic recorders with specifications. $4.95

2768. MAGNETIC RECORDING, Begun

Thoroughly covers the theory of magnetic recording, various types and makes of recorders, their applications and performance measurements. Includes chapter on important research problems. $5.00

2770. HOW TO USE A TAPE RECORDER, Hodgson and Bullen

Written to help business and home recorder owners to learn how to get full value from their machines. Explains basic working of tape recorders and accessories as modern tools of communication. $4.95

See Your Hi-Fi Dealer Or Use This Coupon Today!

Leading hi-fi dealers and salons and radio and electronics parts jobbers are making their stores headquarters for books on every electronics subject. You can take this list to your favorite dealer for immediate purchase.

If your local dealer does not carry books, use the coupon for prompt delivery from ELECTRONICS BOOK SERVICE, on a 7-day free trial basis.

ELECTRONICS BOOK SERVICE, One Park Avenue, New York 16, N. Y.

Please send me the book(s) I have listed below for a FREE 7-day Trial Examination. I understand that if I am not completely satisfied, I may return my selection(s) and I'll owe you nothing. Otherwise, I will send you payment for the book(s) of my choice, plus postage and handling. 1706

NUMBER	TITLE	PRICE
	*TOTAL	

(If you need more space to list additional titles, attach a sheet of paper with additional list.)

☐ SAVE MONEY! Enclose payment in full for the book(s) of your choice and we will pay shipping charges. Same return privileges and prompt refund guaranteed.

☐ Please send me FREE CATALOG, when published.

*New York City Residents, please add 3% sales tax.

NAME ____________________
PLEASE PRINT CLEARLY

ADDRESS ____________________

CITY ____________ ZONE ______ STATE ____________

Power Output Ratings for Hi-Fi Amplifiers

By **VICTOR ROBINSON**
Chief Eng., PACO Electronics Co., Inc.

The meaning and measurement of steady-state, peak, and music power along with discussion of power-bandwidth.

One of the first steps the prospective purchaser of a hi-fi amplifier undertakes is a comparison of advertised power output ratings. It becomes immediately obvious to him that the higher-priced amplifiers are almost invariably associated with higher output ratings and although there may be a number of apparent irregularities in a "graph" of power output *vs* selling price, the prospective buyer can be quite sure that more watts in any manufacturer's amplifier line will mean more dollars.

The answer to the question "How much output power do I really need?" is complex and involves many factors such as speaker efficiency, room acoustics and size, and maximum permissible distortion. Before any of these more refined considerations can be tackled, the apparently simple wattage specification must be examined and defined, inasmuch as several types of wattage rating expressions are currently being employed by amplifier manufacturers in their advertised specifications.

Any power-output rating must have an implied or stated relationship to a distortion rating. A 20-watt rating for an amplifier carries within itself a distortion limit even if this limit is not clearly defined in the manufacturer's specifications. If a distortion limit is listed, for example, as "no greater than 1% total harmonic distortion" and power is listed as "20 watts," it can be expected that as a standard test signal at the input of the amplifier is increased, the graphical crossover point between rising power and rising distortion will be reached at 1% distortion and 20 watts of output power. This point is an important one inasmuch as relatively low-priced amplifiers can show a considerable power advantage if the distortion limit is specified a bit higher than the figure usually expected. This condition exists in this price class of amplifiers wherein little negative feedback may be employed, producing a gently sloping power *vs* distortion characteristic. Medium- and higher-priced amplifiers, however, usually use appreciable inverse feedback, resulting in a sharp rise in distortion above 1% or 2%. Little advantage is gained here if the maximum distortion limit is specified a bit higher.

When making tests for power output, a non-inductive output load resistance must be substituted for the loudspeaker. This condition for test eliminates the complications which would be introduced by the varied impedance *vs* frequency characteristics of speakers and the various efficiencies and characteristics of speaker systems.

With these thoughts in mind, we can proceed to discuss the four types of expressions for output power commonly used: 1. steady-state (r.m.s.) power; 2. "peak" power; 3. music power (often referred to as music waveform power output, or "music waveforms," or "program material" power, or "short bursts" power), and 4. power-bandwidth output.

Steady-State Power Output

This may be defined as "the power output measured across the load resistor at the output of an amplifier with 1000-cps sine-wave signal impressed at the input of the amplifier." This type of test reveals the power capabilities at 1000 cps only; it does not reveal the performance at either the high- or low-frequency ends. It does subject the amplifier to a relatively heavy duty test cycle because of the fact that a sine-wave cycle has a heavier duty characteristic as compared to the more pulse-like character of music or other types of program material. Because of this latter factor and the fact that this type of test can be easily duplicated through use of a sine-wave generator, a.c. voltmeter, distortion analyzer, or scope, the steady-state output specification is used fairly universally and is considered quite revealing.

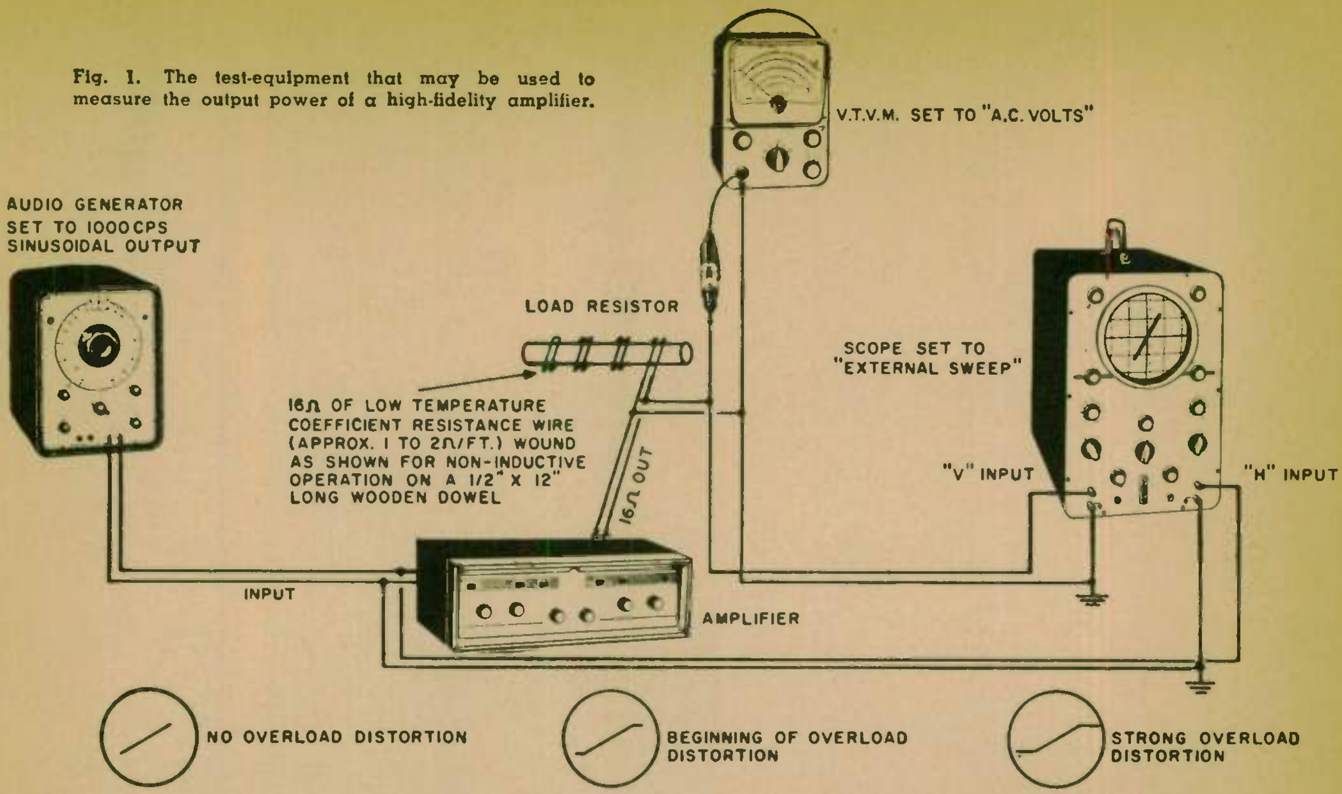
A typical test setup for measuring steady-state power output is shown in Fig. 1.

If the amplifier includes bass and treble controls, these controls are set to the "flat" positions. The signal is increased to the point just below the level wherein clipping begins to occur. By using $P = E^2/R$, the power can be calculated from the voltage reading or by use of the graph shown in Fig. 2. To be as fair as possible to both the amplifier manufacturer and to the amplifier owner, the line voltage should be checked and set, if necessary, to the standard value of 117 volts.

This scope method for checking steady-state power output can be used only to determine *maximum* output values. If power output tests at lower distortion points are desired, a distortion analyzer must be used in place of the scope.

It is important to note that if the

Fig. 1. The test-equipment that may be used to measure the output power of a high-fidelity amplifier.



amplifier spec sheet simply lists "power output" as "10 watts" or "20 watts" and does not further describe the "type" of watts, it can be assumed that the manufacturer is talking about "steady-state (based on r.m.s. values of voltage and current) watts."

Peak-Power Output

This expression in no way relates to the concept of peaks of program material. It is merely a convenient mathematical conversion of r.m.s. power to the a.c. expression for peak of root-mean-square (r.m.s.). This conversion reveals that for a sine-wave signal *Peak Power* = 2 × *r.m.s. Power*, derived as follows:

$P_{rms} = \frac{E^2_{rms}}{R}$, also $1.414 \times E_{rms} = E_{Peak}$ or $\sqrt{2} \times E_{rms} = E_{Peak}$

$$P_{rms} = \frac{E_{rms} \times E_{rms}}{R}$$

$$P_{Peak} = \frac{\sqrt{2} \times E_{rms} \times \sqrt{2} \times E_{rms}}{R}$$

$$P_{Peak} = \frac{\sqrt{2} \times \sqrt{2} \times E^2_{rms}}{R}$$

$$P_{Peak} = \frac{2E^2_{rms}}{R} = 2\left(\frac{E^2_{rms}}{R}\right)$$

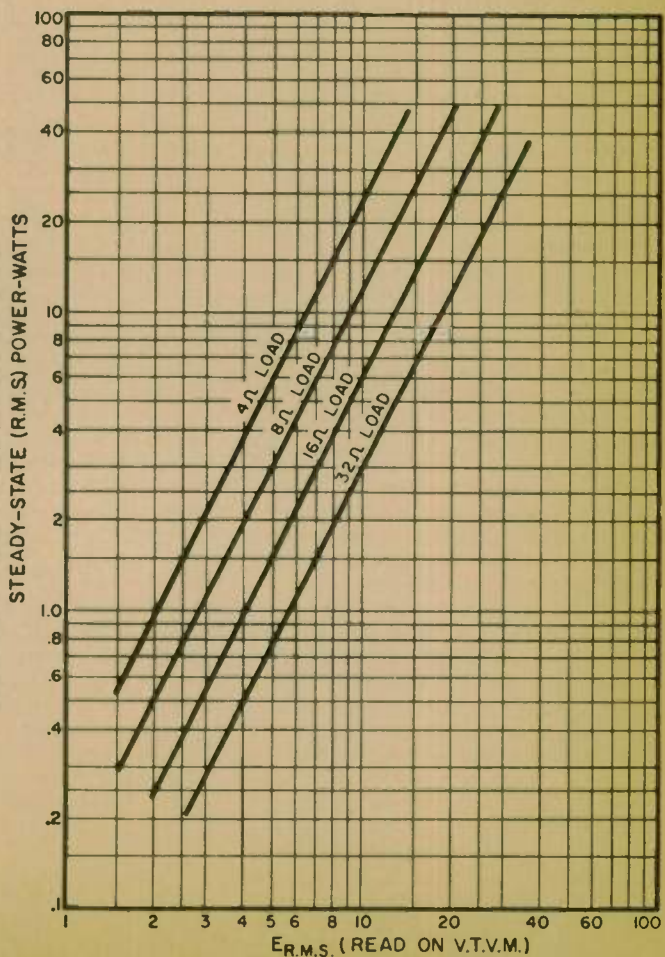
Therefore: $P_{Peak} = 2 \times P_{rms}$

Steady-state (r.m.s.) power can be conveniently measured and compared. Peak power *could* be measured through use of peak-indicating voltmeters, but it is somewhat easier to obtain peak power simply by multiplying r.m.s. power by 2. For practical purposes, "peak power" is not significant in evaluating amplifier performance. It is unfortunate that the term "peak" in this commonly used expression infers a "maximum" rating, *apparently* relating to peak *program* conditions, when actually such is not the case.

Music Power Output

In an attempt to standardize on a power output rating which bears a reasonable relationship to an ampli-

Fig. 2. The graph at the right shows the r.m.s. voltages as read on an output meter or a v.t.v.m. for various power output levels. Values are shown for four commonly used output impedances. As an example of the use of the graph, assume you are measuring the voltage across an 8-ohm load resistor connected to the 8-ohm output terminals. If the r.m.s. voltage reading is 5 volts, the steady-state power output is just a bit over 3 watts as indicated by graph.



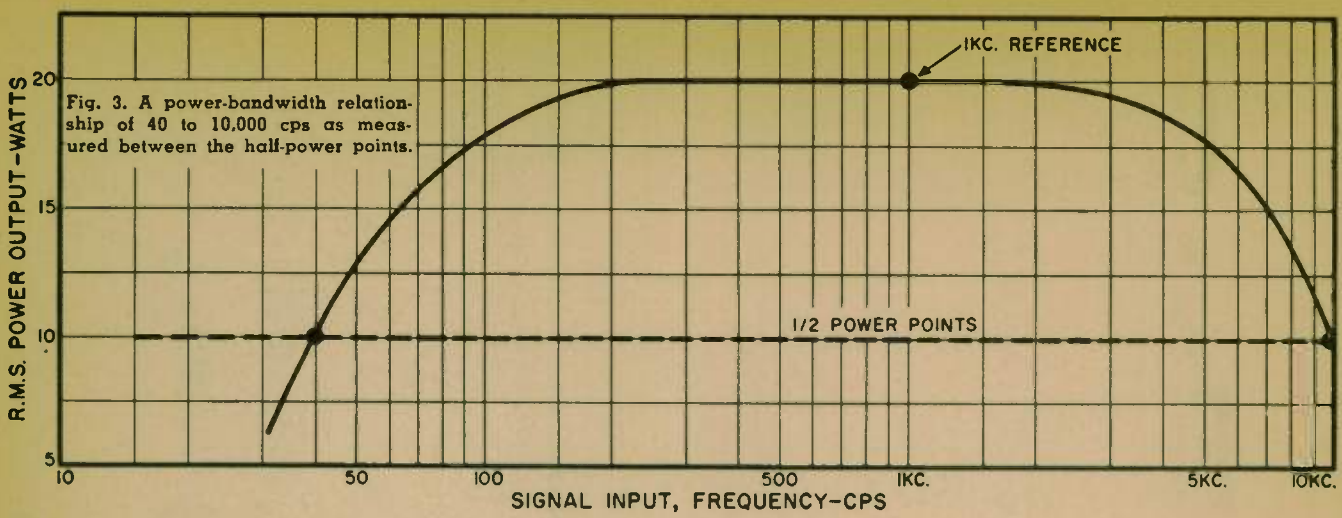


Fig. 3. A power-bandwidth relationship of 40 to 10,000 cps as measured between the half-power points.

fier's capability for reproducing musical program material at high levels, the term "music power" has been developed. The term has been defined as the maximum amplifier power delivered to a resistive load as a result of applying a short-duration 1000-cps sine-wave input signal. The short-duration signal is specified to establish the requirement that output tube potentials must remain unchanged for "no-signal" conditions to "full-signal" conditions. The maximum power rating (music power) under these conditions is limited by the maximum distortion figure specified for the amplifier. The most significant aspect of these test conditions is the requirement that output tube potentials at maximum-output conditions remain up at the "no-signal" value. This requirement stems from an effort to make this "music power" spec simulate the amplification of typical music.

Because of the basic complex character of music, the *average* dynamic range of musical material is relatively low as compared to *peaks* caused by percussion instruments such as the piano, xylophone, etc., for example. The amplifier power required to reproduce the *average* level is correspondingly low; *generous* power, however, is required to reproduce musical *peaks* with low distortion.

With *no* signal applied to the input of the amplifier, the plate and screen voltages of the amplifier are at "no-load" values. As soon as signal is applied, screen and plate currents increase. If the amplifier's power supply has poor regulation, the tube potentials at high signal levels (peak music waveforms) will drop considerably. This voltage drop reduces the low-distortion power capabilities during these peak musical passages and this may produce waveform clipping (distortion) resulting in "muddy" reproduction. The better the regulation of the power supply, the less the difference between "no-signal" and "peak signal" power-supply voltages and the higher the small-distortion power capabilities of the amplifier. The ideal case is achieved with near-perfect regulation, wherein the "peak-signal" power-supply voltage is practically the same as the "no-signal" supply voltage.

If the program or musical peak is of very short duration, however, the energy storing capabilities of the power supply's filter capacitor maintains the tube potentials at practically "no-signal" value (light load) even in the case of poorly regulated power supplies. On *long*-duration musical peaks, however, the voltage produced by a poorly regulated power supply begins to drop, producing greater distortion at the end of the musical peak.

To make "music power" a practical yardstick, it must be assumed that most musical peaks are of *short* duration. If this assumption is acceptable, then the regulation of the power supply can be assumed to be perfect.

Inasmuch as no power supply has perfect regulation it is necessary to simulate perfect regulation in order to perform a *test* for music power. Theoretically, the test can be performed by starting with no signal and then impressing a 1000-cps signal of such short duration that it will produce no drop in tube potentials. In practice, this type of test is difficult to perform with standard test equipment.

For practical purposes, it is simpler to disconnect the amplifier's own power supply and connect an external power supply with controlled output to plate and screen circuits. As a 1000-cps signal of *any* duration is applied to the amplifier's input, the external power supply can be adjusted so that the tube potentials can be held to the "no-signal" values, thereby simulating perfect regulation. Power output calculations made under these conditions will then agree with the conditions set up in the definition of "music power."

Power Bandwidth Output

All of the three preceding power output measurements are made using a standard 1000-cps sine input signal. They obviously do not reveal the power performance at the low and high frequencies. Well engineered amplifiers with generously designed output transformers will usually produce adequate power at the frequency extremes. Occasionally a manufacturer will disclose the high- and low-frequency power performance of his amplifier by using the "power bandwidth" expression which is defined as "those frequencies at which the maximum 1000-cps r.m.s. power output drops to one-half its value" (all readings taken at maximum specified distortion level). For example, if the maximum r.m.s. power output (taken with a 1000-cps sine signal input) is 20 watts and drops to 10 watts as the signal frequency is lowered to 40 cps and also drops to 10 watts as the signal is increased to 10,000 cps, then the "power bandwidth" will be "40 to 10,000 cps". See Fig. 3. Although this type of specification reveals two more pertinent performance characteristics of the amplifier, it is not used by all manufacturers probably because of the possibility of misinterpretation on the part of the reader. *Frequency response* of amplifiers is usually in the range of "20 to 20,000 cps" while the *power bandwidth* spec is almost invariably narrower. Thus an advertised power bandwidth of, for example, "40 to 10,000 cps" might be confused with the frequency response in the mind of the customer.

An understanding of these power output expressions can be a help in analyzing an amplifier on the basis of the manufacturer's specifications as follows: (A) If *both* r.m.s. power *and* music power are listed, the closer the r.m.s. power approaches the music power rating, the better the power supply regulation and the better the maintenance of low distortion on extended power peaks. (B) If the r.m.s. power *only* is listed and it is *sufficiently high* for your purposes, then the value of music power becomes academic inasmuch as the r.m.s. power maximum determines the power capabilities on *extended* power peaks, and (C) if music power is listed and r.m.s. power is *not* listed, then there is a chance that the power-supply regulation may not be as good as in amplifiers which have music power ratings close to the r.m.s. power rating, unless further examination of the amplifier's specifications indicates the incorporation of a low-impedance (well-regulated) power supply.

Compact 20-Watt Hi-Fi Amplifier

By **HAL WITTLINGER**
Electron Tube Div., Radio Corp. of America

Easy-to-build circuit with low distortion, high sensitivity, and voltage regulation.

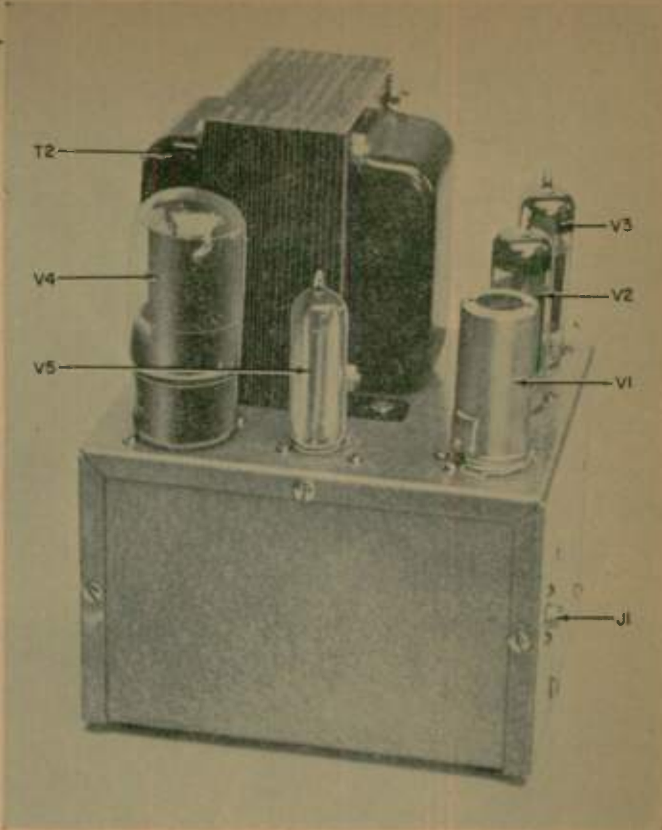
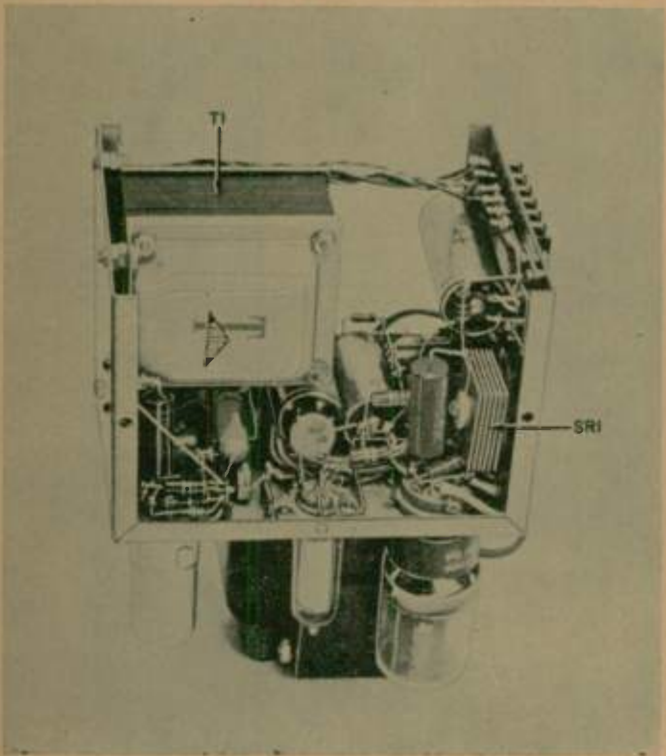
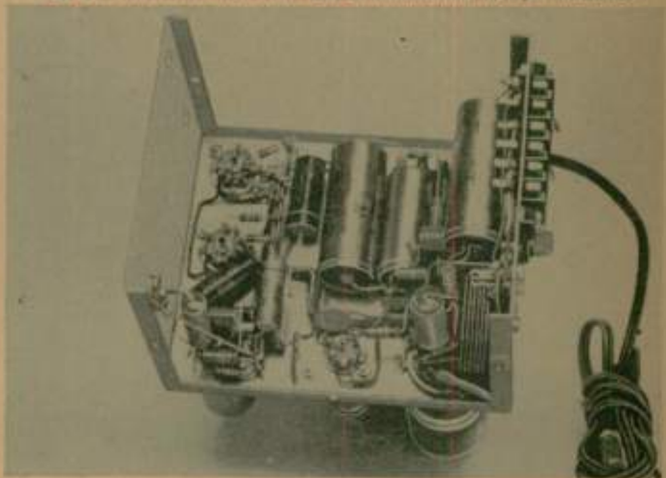


Fig. 1. Complete amplifier is built on 4" x 5" x 6" chassis box.



◀ Fig. 2. Under-side of the amplifier chassis showing components.

Fig. 3. Under-side view with output transformer not yet mounted.



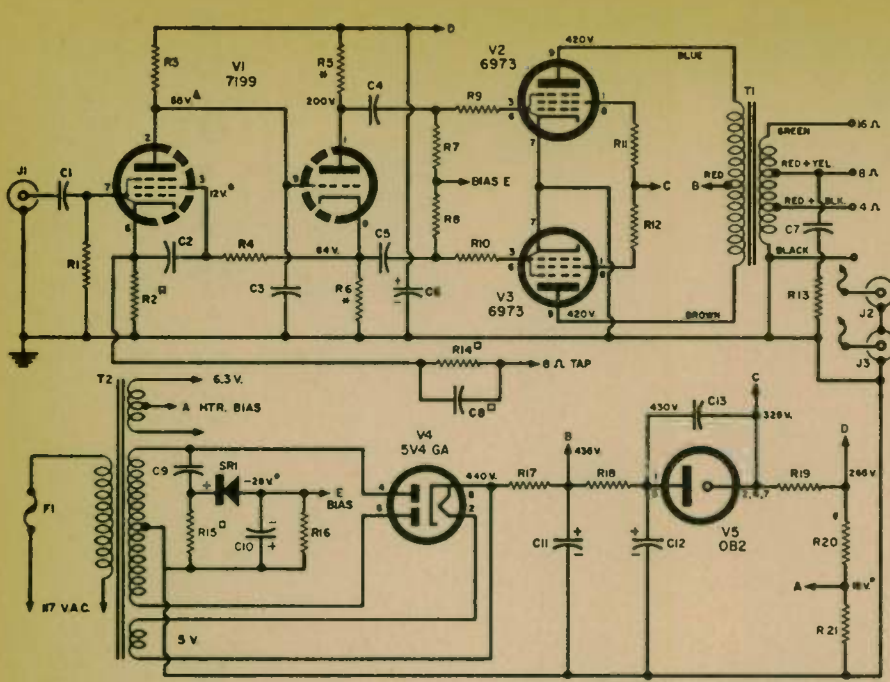
THIS article describes a compact, economical, and easily built audio power amplifier which is equally useful as the basic unit of a high-quality monophonic system or as a second amplifier for stereo conversions. The amplifier uses the recently announced *RCA* "High-Fidelity" tubes and is built complete with power supply on a 4" x 5" x 6" "Minibox" chassis. Despite its small size, it is capable of developing 20 watts with only 0.6% total harmonic distortion and 1.55% intermodulation distortion; has excellent frequency response characteristics; extremely low hum and noise; and high sensitivity.

Circuit Details

Fig. 4 is the circuit diagram and parts list for the amplifier while the photographs of Figs. 1, 2, and 3 show the layout and mechanical construction. An *RCA* 7199 medium-mu triode, sharp-cut-off pentode, which has specially controlled hum and noise characteristics, is used as a high-gain voltage amplifier and phase-splitter, driving push-pull *RCA* 6973 beam power tubes operating in class AB, with fixed bias.

Direct coupling between the voltage-amplifier and phase-splitter stages minimizes phase shifts and permits the use of 22 db of inverse feedback between the voice-coil terminals of the output transformer and the cathode of the input stage. This relatively large amount of feedback helps minimize distortion and hum and provides the low output impedance necessary for effective speaker damping. The output impedance at the 8-ohm voice-coil terminals is 0.8 ohm (damping factor = 10).

The voltage gain of the pentode input stage with the feedback loop open is in excess of 200. The constants of



R_1—330,000 ohm, ½ w. res.
R_2, R_{13}—1000 ohm, ½ w. res.
R_3, R_4—820,000 ohm, ½ w. res.
R_5, R_6—15,000 ohm, 1 w. res. (matched 1%).
R_7, R_8—270,000 ohm, ½ w. res.
R_9, R_{10}—470 ohm, ½ w. res.
R_{11}, R_{12}—100 ohm, ½ w. res.
R_{14}—33,000 ohm, ½ w. res.
R_{15}—5600 ohm, 1 w. res.
R_{16}—56,000 ohm, ½ w. res.
R_{17}—56 ohm, 1 w. res.
R_{18}—470 ohm, 1 w. res.
R_{19}—10,000 ohm, 1 w. res.
R_{20}—150,000 ohm, 1 w. res.
R_{21}—10,000 ohm, ½ w. res.
C_1—.05 µf., 400 v. capacitor
C_2—.5 µf., 200 v. capacitor
C_3—15 µµf. mica capacitor
C_4, C_5—.1 µf., 400 v. capacitor
C_6—20 µf., 350 v. elec. capacitor
C_7, C_{13}—.02 µf., 200 v. capacitor
C_8—82 µµf. mica capacitor
C_9—.05 µf., 600 v. capacitor
C_{10}—100 µf., 50 v. elec. capacitor
C_{11}, C_{12}—40 µf., 450 v. elec. capacitor
J_1, J_2, J_3—Phono jack
SR_1—75 ma. selenium rectifier
F_1—2 amp fuse
T_1—Output trans. 8000 ohms c.t. to 16, 8, 4 ohm v.c. (Triad S-31A)
T_2—Power trans. 350-0-350 v. @ 120 ma.; 5 v. @ 3 amps; 6.3 v. c.t. @ 4.7 amps. (Thordarson 24R05U or equiv.)
V_1—7199 tube (RCA)
V_2, V_3—6973 tube (RCA)
V_4—5V4-GA tube (RCA)
V_5—0B2 tube

Fig. 4. Circuit of the miniature high-fidelity amplifier, with typical voltages.

this stage and of the direct-coupled phase-splitter stage were carefully chosen to assure class A operation of these stages at signal levels well beyond that required for full output. Grid-No. 2 voltage for the pentode unit is obtained from the cathode of the triode unit. Because of the large time-constant of the grid No. 2 circuit (0.82 megohm x 0.5 µf.), the grid-No. 2-to-cathode potential is essentially constant for audio frequencies. This arrangement provides feedback between the two units which stabilizes the operating point of the phase-splitter against changes in the "B" supply voltage and tube characteristics.

Two networks are used to assure stability at high frequencies. The 15-µµf. capacitor connected from the plate of the pentode amplifier to ground provides a gradual roll-off in response beginning at 15 kc. The .02-µf. capacitor and the 1000-ohm resistor across the 8-ohm winding of the output transformer suppress high-frequency oscillations under overload conditions.

Power-Supply Details

The excellent performance of the amplifier is partly due to the design of the power-supply circuit. An *RCA* 5V4-GA rectifier tube is used because of its small internal voltage drop and excellent regulation. Its indirectly heated cathode also delays the build up of full "B" supply voltage until the tubes have reached operating temperature, thus minimizing voltage surges across the power-supply components. Grid-No. 2 voltage for the 6973 output tube is obtained from the plate-supply voltage through an 0B2 glow-discharge voltage-regulator tube. This arrangement provides the extremely stable grid-No. 2 voltage necessary to assure low distortion at full output and minimizes the current drain on the power transformer because it eliminates the need for power-wasting, voltage-dropping resistors or bleeder networks.

Construction

Fig. 1 is a top view of the amplifier while Figs. 2 and 3 show the underside and wiring. With the exception of the relative positions of the power and output transformers, placement of components is not critical.

No marked effect on the amplifier's stability or performance was noted despite the fact that it was built in the rather limited confines of the 4" x 5" x 6" "Minibox." The windings of the output transformer are well shielded electrostatically by the core and the transformer housing. Coupling from the magnetic field of the output transformer is not a problem because this field is of very small magnitude. To minimize undesired coupling from the plate lead of the output transformer, the transformer was mounted so that these leads come out of the case on the side farthest from the input stage.

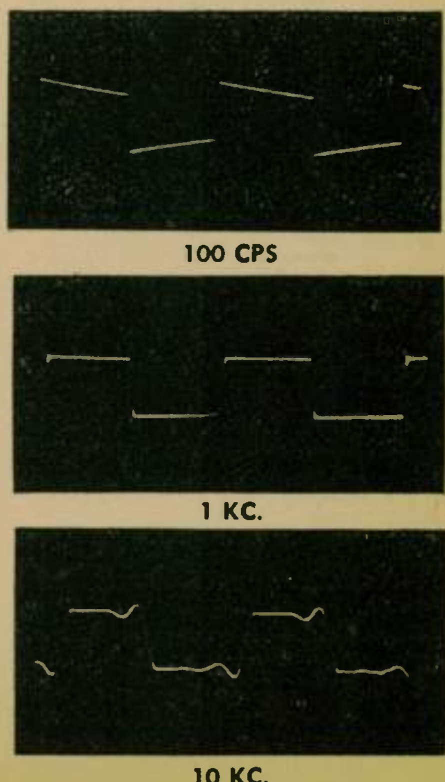
Because of its proximity to the power transformer, the output transformer picks up a small a.c. component. The resulting hum has a level of about 100 db below 20 watts and can be reduced, if desired, by rotating the output transformer 5 to 10 degrees while observing the a.c. output of the amplifier on an oscilloscope or a high-sensitivity voltmeter.

Compact construction often engenders heat-dissipation problems. In this amplifier the principal sources of heat, *i.e.*, the tubes and power transformer, are mounted on top of the chassis so that the heat rises without detrimental effect on the components on the underside of the chassis.

To increase the flexibility of this amplifier, two phono jacks are mounted on the rear of the chassis and connected to two spare terminals on the output-terminal strip. Any tap on the voice-coil winding can thus be connected to either jack by connecting a jumper between the appropriate binding posts.

The following suggestions will help

Fig. 5. Square-wave response of amplifier.



you obtain maximum performance from this or any other audio amplifier you construct.

1. Dress the a.f. input lead away from the heater wiring. Hum picked up in the a.f. input circuit is not reduced by feedback, because this circuit is outside the feedback loop.

2. Ground the "B-minus" bus to the chassis only at the a.f. input terminal. If a plug-and-socket arrangement is used for the speaker connection, be sure the ground side of the voice-coil circuit is connected only to the "B-minus" bus and not to the chassis of the amplifier.

3. Connect all ground returns to the "B-minus" bus in the order shown on the schematic. Remember, the high ripple current in the plate supply for the output stage can raise the amplifier's hum level if it enters either of the preceding stages through the common "B-minus" bus. Although this type of hum originates within the amplifier's feedback loop, it often becomes so great that it cannot be eliminated by feedback.

Performance Data

Fig. 6B shows the frequency response curve of the amplifier while Fig. 6A shows its total harmonic and intermodulation distortion as functions of power output. Note that at normal home-listening levels both IM and harmonic distortion are substantially below the one-half of one per-cent level. Despite the relatively large amount of inverse feedback used the amplifier has a sensitivity of .42 volt r.m.s. for 20 watts output. Hum and noise, with input shorted, are 84 db below 20 watts.

Fig. 5 shows the square-wave response of the amplifier at 100, 1000, and 10,000 cps. The slight overshoot in the 10,000-cps square-wave response is due to the peak in the 20-watt frequency-response characteristic at 48,000 cps (see Fig. 6B).

As a check on the performance of the input-amplifier and phase-splitter stages, the combined cathode currents of these stages were measured over a wide range of signal levels. No change in current occurred until the output of the phase splitter reached a level of 170 volts peak-to-peak, indicating that there is no departure from class A operation up to signal levels three times greater than that required to drive the 6973's to full output.

Because a major factor in the performance of an audio power amplifier is the output transformer, considerable care was exercised in selecting a compact transformer which would provide the desired performance yet be moderate in cost. Several small output transformers having suitable power ratings were tried, but these lacked the necessary response and power-output capabilities at extreme high and low frequencies. Although the transformer used is not the most expensive small unit available, it proved to be an excellent choice for this compact economy amplifier.

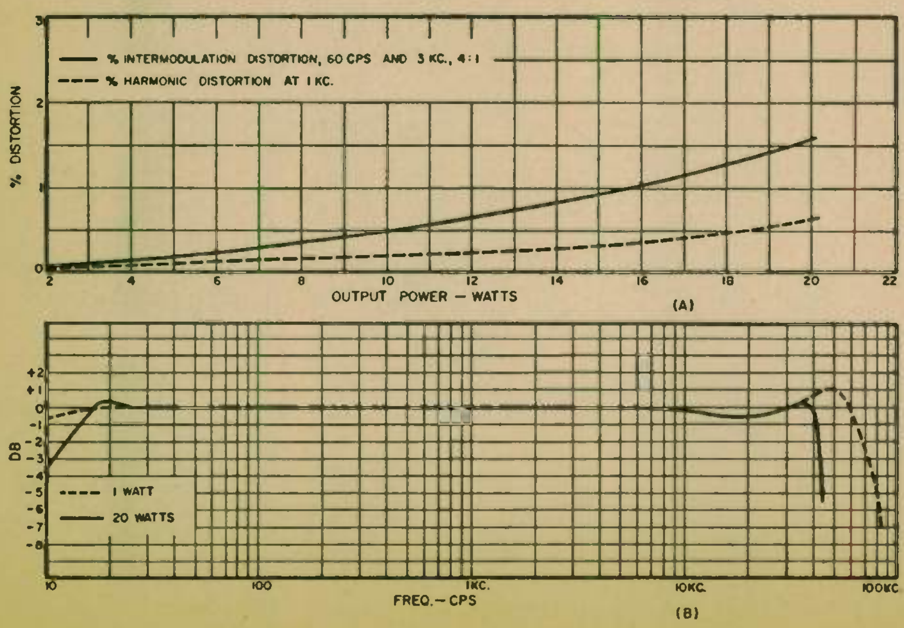
Power Output	20 watts
Sensitivity	0.42 volt input for 20 watts
Hum & Noise	84 db below 20 watts
Distortion	Total harmonic: @ 1 kc., 20 w. = 0.6%; IM @ 60 cps & 3 kc. (4:1) @ 20 w. = 1.55%
Damping Factor	10 @ 60 cps, 8-ohm tap

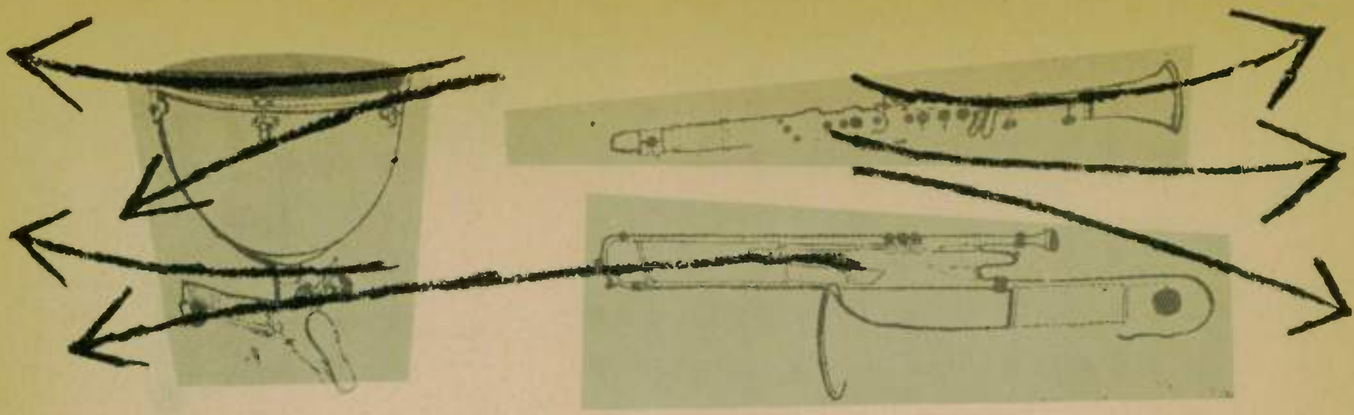
Table 1. Important specifications of unit.

Although the output transformer is rated at 15 watts, it performs well at the 20-watt level. Because one rarely operates an amplifier continuously in the home at more than a few watts output, there should be little danger that the manufacturer's ratings will be exceeded.

Observe the precautions listed when you build this amplifier and you will have a unit that provides many enjoyable hours of monophonic or stereo listening.

Fig. 6. (A) Intermodulation and harmonic distortion. (B) Frequency response.





Bass-Treble Hi-Fi Amplifier

By FRANK R. BARKEY

Complete construction details on a power amplifier that delivers about 20 watts of bass power and 20 watts of treble power to separate speaker systems.

THE possibilities for improved reproduction through the use of two channels seem to have been neglected. Comparatively few double-channel amplifiers are on the market today for home use. There are, of course, those who use two amplifiers to feed separate speakers but even these are relatively few and their main interest is with stereophonic reproduction.

The potential ability of two channels to give better transient reproduction, lower harmonic and intermodulation distortion, cleaner bass and greater tonal separation has been overlooked seemingly because of several faults which are not impossible of correction. The question of cost does not seem a factor since the outlay for an additional output transformer and several tubes is balanced by a simpler preamplifier and elimination of the speaker crossover. Less expensive output transformers may give good results.

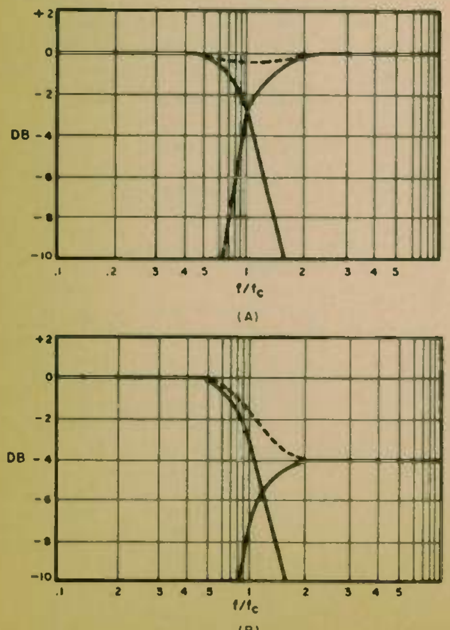
Two difficulties arise in the use of twin amplifiers or twin channels. They result, in part, from the fact that a good amplifier is designed for flat, wide-band response and partly from the necessary use of an electronic crossover. The first difficulty shows up when we analyze the combined output from two channels. Fig. 1A shows the type of response curve usually associated with this type of operation. There is flat bass response with sharp roll-off at the crossover frequency. Similarly the treble response is flat with an equally abrupt roll-off. The combined response is excellent until we change the setting of the gain control in either channel. Such change creates a sharp step in the over-all response, either up or down, at the crossover frequency, as shown in Fig. 1B.

The second difficulty is the large amount of phase shift occurring at the crossover network. It is usual to use two, sometimes three, *RC* stages in each leg to secure the desired roll-off.

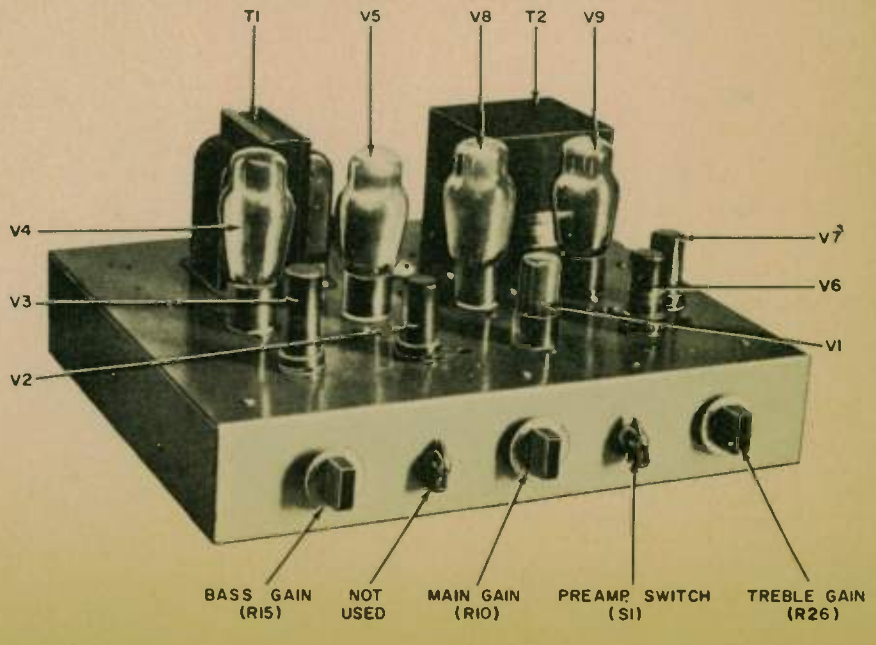
The solution to these two problems provides an opportunity for improved reproduction which is not possible with single-channel amplifiers, nor indeed with twin amplifiers.

Good frequency response doesn't necessarily mean good transient reproduction any more than good frequency response means high-quality reproduction. Most single-channel amplifiers, and twin-amplifier arrangements, cause distortion of transients to a greater or lesser degree. The main reason, per-

Fig. 1. Usual two-channel output curves with (A) similar and (B) different gains.



Over-all view of two-channel amplifier showing location of controls and tubes. The bass portion of the amplifier is on the left, the treble is on the right.



haps, why one amplifier sounds cleaner and has more of that elusive "presence" is because of better transient reproduction. When speaking of transients this writer refers to any sound, musical or otherwise, which is not a continuous tone. By this definition the piano produces transient sounds only, in contrast to the violin or the wind instruments which produce continuous tones

Good transient reproduction is not as simple as the reproduction of continuous-state sounds. The audible quality of a steady tone doesn't suffer too much from moderate degrees of phase shift, nor is it objectionably degraded by changes in the phase relationship of its component frequencies. About the worst that can be expected is a change in timbre. Transients, on the other hand, are greatly affected by phase changes. A transient waveform is complicated. Its component frequencies may be unrelated harmonically and spaced widely. The shape of the waveform, and hence its sound, is delicately dependent on the phase relationship of the component frequencies. Any change in this relationship alters the transient waveform and it may sound different.

To illustrate how easily this occurs, let us assume a transient containing frequencies of 50, 500, and 5000 cycles per second. Across a simple coupling circuit such as Fig. 2, phase shift can be expressed: $\tan\phi = \frac{1}{\omega CR}$.

At 50 cycles: $\tan\phi = \frac{1}{\omega CR}$

$$= \frac{1}{2\pi \times 50 \times .05 \times 10^{-6} \times .5 \times 10^{6}}$$

$$= \frac{1}{\pi \times 2.5}$$

$$= 7^\circ \text{ of phase shift}$$

Over three similar circuits in the average amplifier this totals 21° of phase shift at 50 cycles. At 500 cycles the total will equal roughly 1°. At 5000 cycles it is 0° for all practical purposes. Oversimplified as this is it can be seen that we have altered the shape of the transient waveform. Note, however, that above 500 cycles, the 10th harmonic, there is practically no change (less than 1°) in the original phase relationship. It follows, then, that if phase shift at the fundamental and its 10th harmonic were made equal we could preserve the original phase relationship at all component frequencies within small limits. This cannot be done in a single-channel amplifier, nor in a twin-amplifier arrangement, except by using impractically large values of C in the coupling circuits. We can, however, design one channel to reproduce low frequencies with minimum phase shift and a second channel which will duplicate this shift at the 10th harmonic. In other words, low-channel phase shift at 100 cycles will equal high-channel shift at 1000 cycles.

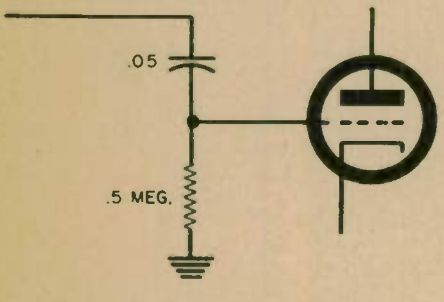
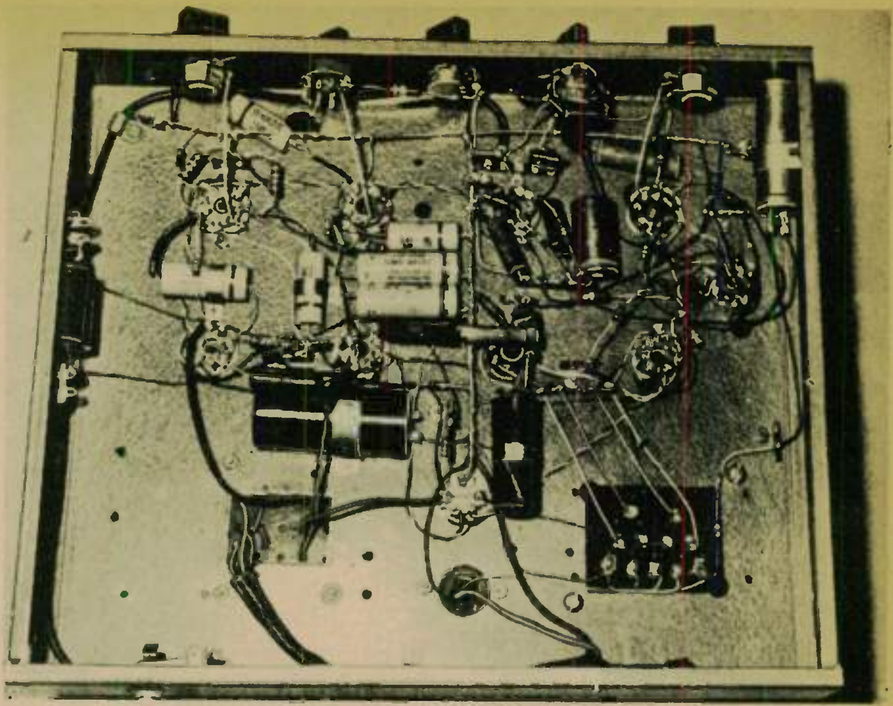


Fig. 2. Simple RC coupling circuit.



Underside of chassis. A heavy ground bus, supported on insulated tie points, runs through the center of the chassis and across the front. It is grounded to the chassis at only one point near the input. Between the output transformers is the power socket, below which is a rubber-lined opening to the output terminal strip.

The problem of large over-all phase shift at the crossover network can be minimized by using a single-stage network. Moderate degrees of shift will not affect the sound of the transient *providing* we maintain the phase relationship of the component frequencies. A single-stage crossover can be designed which will: (1) equalize high- and low-channel phase shift, (2) alter the response curves to eliminate frequency distortion caused by the step of Fig 1B.

Two response curves are shown in Fig. 3. The bass curve has a constantly rising response at approximately 6 db per octave The treble curve has flat response with a slow roll-off at the low-frequency end. If these were the characteristics of two channels, changing the setting of the gain controls would shift these curves up or down in relation to each other, but at no usable crossover frequency would we obtain the configuration of Fig. 1B. The gain controls would do three things: (1) adjust the balance of highs and lows, (2) set the crossover frequency, and (3) provide continuous control of the crossover frequency. With the amount of bass boost indicated there would be little need for boost in the preamplifier which might even be a simple voltage amplifier, without compensation.

Circuit Used

The circuit of an amplifier which incorporates these specifications is shown in Fig. 4. A simple 6SL7 preamp with a switched feedback loop is fed to a single-stage crossover network. Other preamplifiers may be used, of course, but for best results with most LP's it should be possible to switch out the fre-

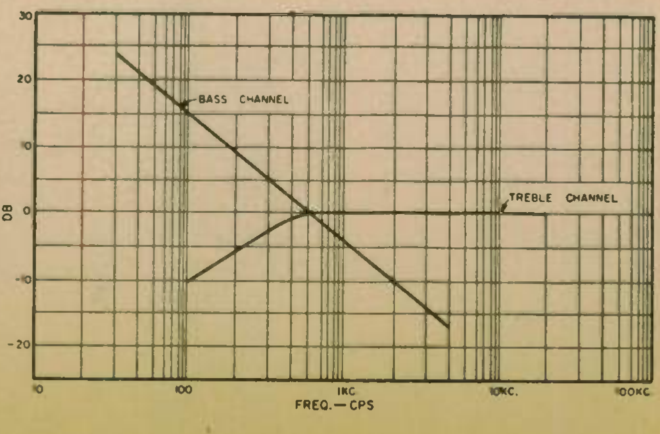


Fig. 3. The idealized bass and treble response curves that would produce the effects described by the author. The actual curves produced by the amplifier are quite close to these idealized curves. (Refer to Fig. 5.)

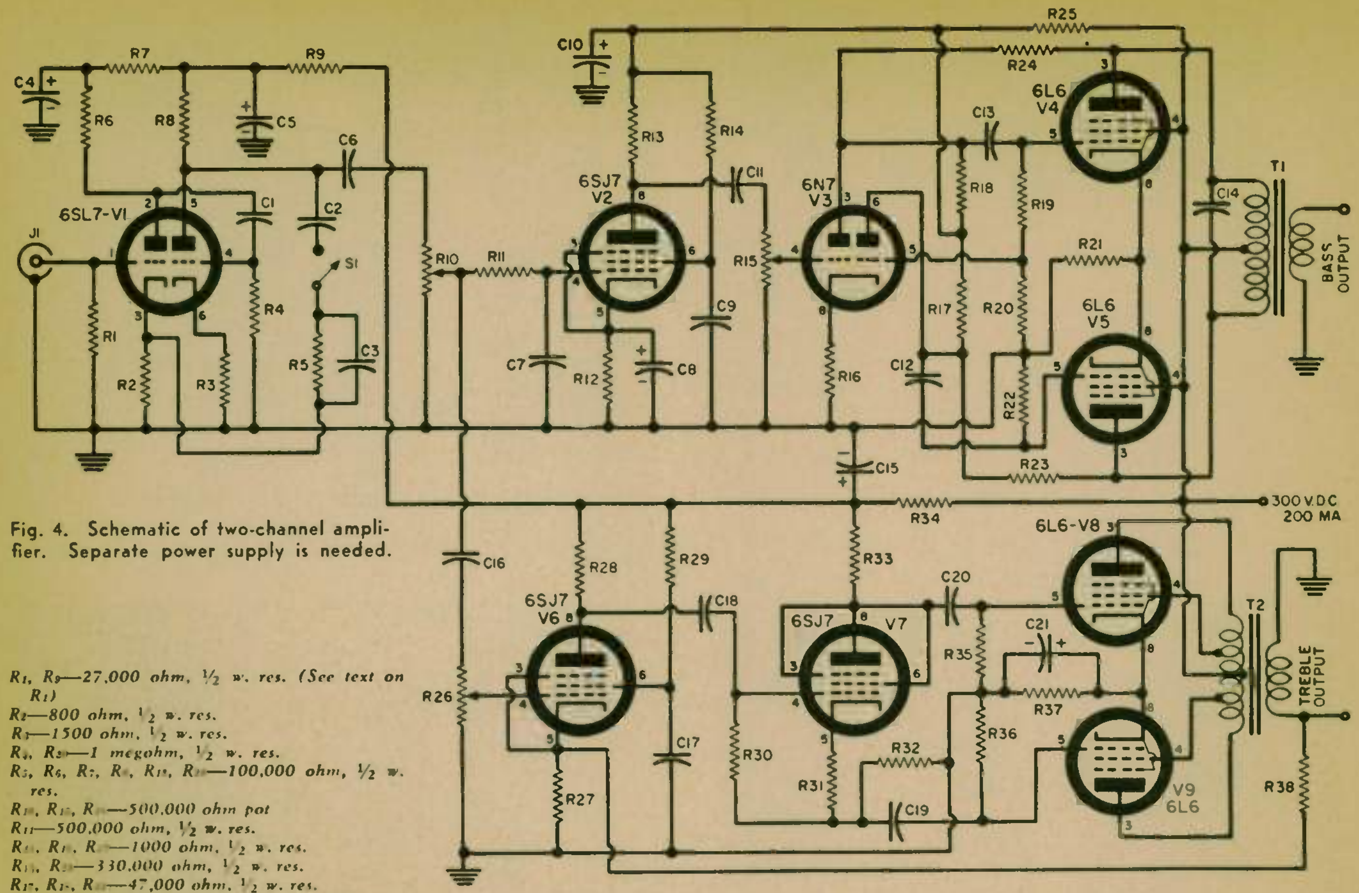


Fig. 4. Schematic of two-channel amplifier. Separate power supply is needed.

R_1, R_9—27,000 ohm, ½ w. res. (See text on R_1)
R_2—800 ohm, ½ w. res.
R_3—1500 ohm, ½ w. res.
R_4, R—1 megohm, ½ w. res.
R_5, R_6, R_7, R, R, R—100,000 ohm, ½ w. res.
R, R, R—500,000 ohm pot
R_{11}—500,000 ohm, ½ w. res.
R, R, R—1000 ohm, ½ w. res.
R, R—330,000 ohm, ½ w. res.
R, R, R—47,000 ohm, ½ w. res.
R, R_{22}, R, R, R, R—270,000 ohm, ½ w. res.
R—21,000 ohm, ½ w. res.
R, R—250 ohm, 10 w. wirewound res.
R, R—10,000 ohm, ½ w. res.
R—2700 ohm, ½ w. res.
R—51,000 ohm, ½ w. res.
R—See text
C_1, C—.05 μf., 400 v. capacitor
C—.002 μf., 400 v. capacitor
C—.0004 μf., 400 v. capacitor
C, C, C, C—20 μf., 400 v. elec. capacitor
C, C, C_{11}, C, C_{13}, C—.1 μf., 400 v. capacitor
C_7, C, C, C—.01 μf., 400 v. capacitor
C—100 μf., 15 v. elec. capacitor
C—.001 μf., 400 v. capacitor
C—20 μf., 50 v. elec. capacitor
J_1—Phono jack
S_1—S.p.s.t. switch
T_1, T—5000 ohms plate-to-plate to voice coil, 20 watts or more (Author used 6600-ohm "Ultra-Linear" type transformer for T_2, see text)
V_1—6SL7 tube
V, V, V_7—6SJ7 tube
V—6N7 tube
V, V, V, V—6L6 tube

quency-compensating networks. The feedback loop shown is used mainly to correct old 78's and reduce surface noise. Fig. 5 shows the response of each amplifier channel. Bass response rises steadily to a maximum of 28 db of boost at 40 cycles (with respect to a reference frequency of 1000 cycles) then rolls off to 24 db at 20 cycles. The treble channel falls off slowly below 500 cycles and is flat ± 1.5 db from 500 cycles to 100 kc. (See Fig. 6)

Neglecting the preamp for the moment, let us assume any fundamental and its 10th harmonic, say 100 cycles and 1000 cycles. Across R_{11}-C_7, the bass side of the crossover, phase shift will be expressed:

$$\tan\phi = \frac{1}{\omega CR}$$

$$= \frac{1}{2\pi \times 100 \times .01 \times 10^{-6} \times .5 \times 10^{6}}$$

$$= \frac{1}{\pi}$$

$$= 17^\circ \text{ of phase shift.}$$

Across the bass channel coupling circuits there will be an additional 6° of shift for a total of 23° at the grids of the output stage.

Across R_{26}-C_{16}, the treble side, at 1000 cycles:

$$\tan\phi = \frac{1}{\omega CR}$$

$$= \frac{1}{2\pi \times 10^{3} \times .001 \times 10^{-6} \times .5 \times 10^{6}}$$

$$= \frac{1}{\pi} = 17^\circ \text{ of phase shift}$$

Across the treble-channel coupling circuits there will be an additional 5° of shift for a total of 22° at the grids of the output stage. The difference of 1° over-all is small enough to be neglected. Although these calculations are oversimplified they seem to work out in practice. Oscilloscope measurements across the voice coils of both channels reveal that phase shift at the bass channel fundamental and at the 10th harmonic in the treble channel are the same within very narrow limits. Since this is so, the transient waveform, or any waveform for that matter, is reproduced with greater fidelity.

Examination of the method used to compensate the recording curve is interesting. The RIAA curve is, at the low end, flat to 500 cycles then rolls off at 6 db per octave to −20 db at 50 cycles. Fig. 5 shows that with a crossover frequency at 500 cycles there is almost exactly 20 db of boost at 50 cycles. We therefore can feed an uncompensated signal to the bass channel for a flat output. At the high-frequency end, the RIAA curve is flat to 2120 cycles then rises at 6 db per octave. At 10,000 cycles it is up 13 db. At normal listening levels this is just the boost required to conform to Fletcher-Munson listening curves. This, of course, is an argumentative point. Individual hearing ability progressively deteriorates with age. Not only is there likely to be a general loss of acuity over the entire frequency range but the ability to hear high-frequency sounds continuously decreases as age advances. For example, representative values of the falling off in ear sensitivity with age, at 4000 cycles, (compared to age 25) are: age 35, 10 db; age 45, 12 db; age 55, 24 db. (*Jensen* Technical Monograph #1, 1944). On the average, however, the 13 db of boost at 10,000 cycles seems to work out satisfactorily for most age groups.

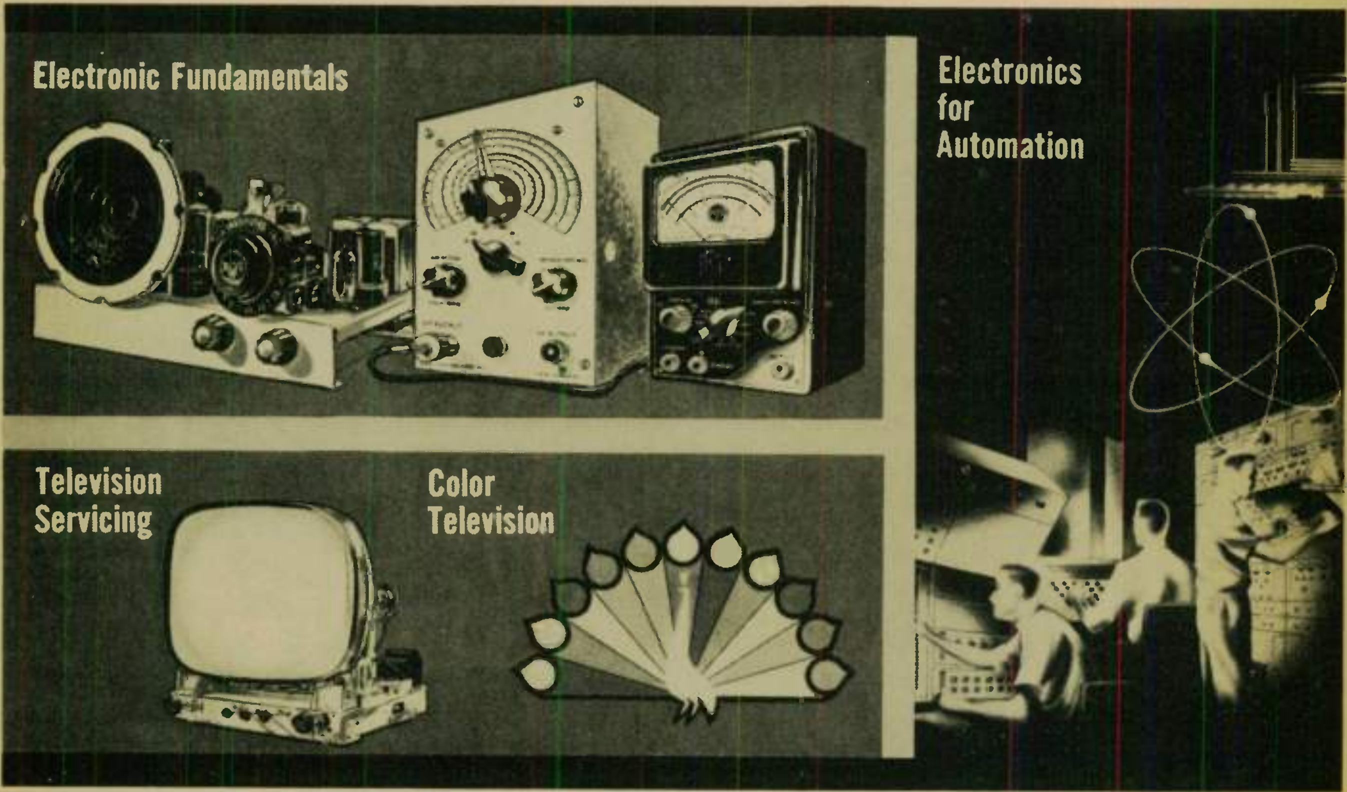
The bass and treble response curves permit useful crossover frequencies from about 150 cycles to 2000 cycles. Setting the correct crossover frequency is not the problem it might appear at first glance. Repeated tests by the writer in setting the crossover point by ear show an average error of ± 20 cycles. The writer, a motion picture technician for over 20 years, happens to have a sensitive and trained ear. For the average listener this degree of accuracy is not probable. However, that

LET RCA TRAIN YOU IN ELECTRONICS

RCA Institutes, one of the world's leading electronic technical schools, offers a Home Study Course in . . .

ELECTRONICS FOR AUTOMATION

. . . Now you have *four* comprehensive courses for your electronic training . . . from basic electronic theory to the more advanced principles of color TV and Automation.



Practical work with the very first lesson. Pay-as-you-learn. You need pay for only one study group at a time.

Send for our 64 page Home Study Catalog

FREE!

RESIDENT SCHOOLS in New York City and Los Angeles offer Technical Institute and Vocational School Courses in Electronics. Day and Evening classes start 4 times each year. Resident School Catalog sent free on request.

RCA INSTITUTES, Inc. Home Study School, Dept. AY-90
A Service of Radio Corporation of America
350 West Fourth Street, New York 14, N. Y.

Please rush me your FREE illustrated 64-page book "Your Career in Electronics," describing your home training programs. No obligation. No salesman will call.

Name . . . . Age . . . .
Please print
Address . . . .
City . . . . Zone . . . . State . . . .
Veterans: Enter discharge date . . . .

CANADIANS — Take advantage of these same RCA courses at no additional cost. No postage, no customs, no delay. Send coupon to:
RCA Victor Company, Ltd., 5581 Royalmount Ave., Montreal 9, Quebec
To save time, paste coupon on postcard.

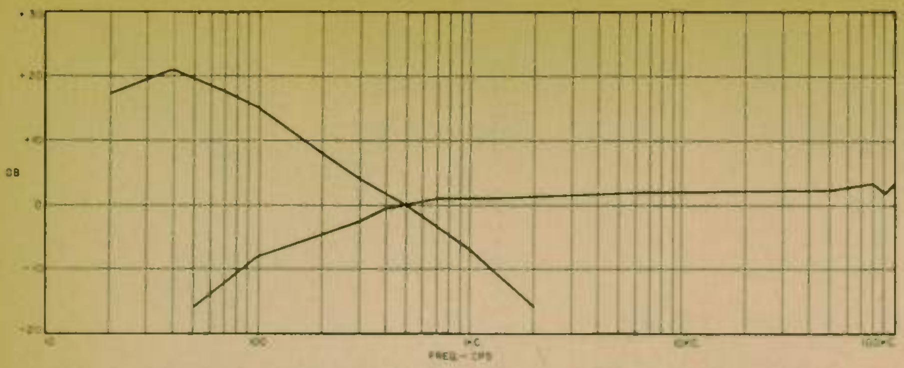


Fig. 5. Response measured across loudspeaker voice coils for each of the channels. Under these conditions the effects of speaker and enclosure resonances are included.

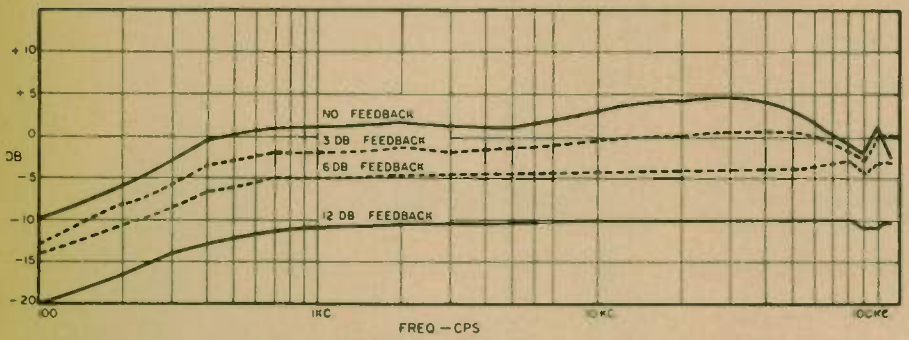


Fig. 6. Measured frequency response across the treble loudspeaker with various amounts of negative feedback employed. "Ultra-liner" type connection was used.

isn't the point. What *is* important is the fact that the listener can obtain a balance most pleasing to his own ears. There is a simple method of doing this. Begin with a recording whose characteristics are unknown and start with both gain controls turned off. Advance the bass control until a comfortable level is obtained. The effect is that of listening through a heavy door. Then simply open the door to the recording studio, advancing the treble control until it sounds as though you, the listener, were in the studio.

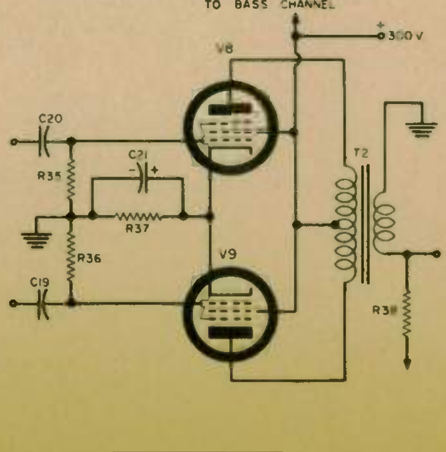
One characteristic of the bass channel is of note in the light of Howard F. Hume's experiments in stereo sound reproduction. (*Audio*, March 1957.) In his experiment #7 he found that the human head is so shaped that it appears to strip harmonics from fundamentals below 800 cycles without affecting the amplitude. The head would seem to act as a type of filter and this may be the reason why distortion is usually less noticeable at low frequencies. In our bass channel, at 70 cycles square waves approach sine-wave shape and square waves above 100 cycles are reproduced as sine waves. As a consequence the amplifier has exceptional bass quality. The pedal line of an organ, for example, is remarkably clean and pure toned.

Since there are few harmonics reproduced in the bass channel, harmonic distortion becomes much less of a problem than it is with single-channel amplifiers. Also, intermodulation products are greatly reduced since large amplitude bass frequencies are completely separated from the higher frequencies. Although no distortion measurements were made, total distortion appears to be less than that caused by the phono pickup that was utilized.

Construction

Points to be considered in construction are not complicated beyond normal care in layout and wiring to minimize hum and obtain optimum balance. From the photographs it will be noted that components are given plenty of room and that wiring has been kept to a minimum. Point-to-point technique, favored in television circuits, has been used as much as possible. This helps to reduce stray capacities and aids stability at high frequencies. The value of R_1 will vary and should be the value recommended for the pickup used. Load resistors in the bass-channel phase-inverter stage should be matched and also the grid resistors of both push-pull output stages. Signal voltages at the output grids should be balanced using v.t.v.m. or oscilloscope. Any unbalance on the bass side can be corrected by changing the value of R_{30} and with the values shown for R_{31}, R_{32}, and R_{33} there should be no unbalance on the treble side. Use of different phase-inverter stages was based on two points: keeping the treble channel as simple as possible and avoiding the hum problem of a high-potential cathode in the bass side. Since hum is not reproduced on the treble side, the high-potential cathode of the inverter stage is not a problem.

Fig. 7. Treble output circuit converted to straight tetrode connection.



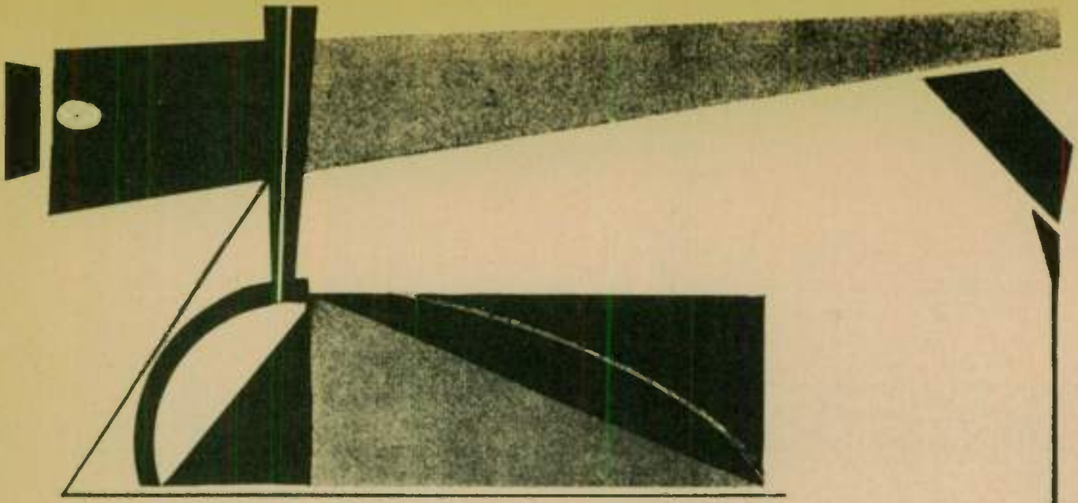
Feedback in the bass channel is used around one stage only to obtain the desired slope in the bass response curve. In the treble channel the value of the feedback resistor will depend, to some extent, on the characteristics of the output transformer used. The author uses an "Ultra-Linear" unit with primary impedance 6600 ohms; primary inductance 50 henrys; leakage inductance 20 millihenrys. The advantages of "Ultra-Linear" operation for the higher frequencies only, however, are not too great. Use of a normal tetrode connection, Fig. 7, will give excellent results at lower cost. Various values of R_{38} give feedback as follows: 330,000 ohms—3 db; 16,000 ohms—6 db; and 3900 ohms—12 db. The effect of feedback in improving frequency response is shown in Fig. 6. It can be seen that more than 6 db of feedback is not warranted. If more feedback is desired, the slight reduction in distortion percentage has to be balanced against the resultant loss of gain. More than 12 db of feedback will require the addition of a push-pull driver stage after the phase inverter. Maximum power output of each channel is approximately 20 watts giving a total power output of 40 watts.

Power Supply

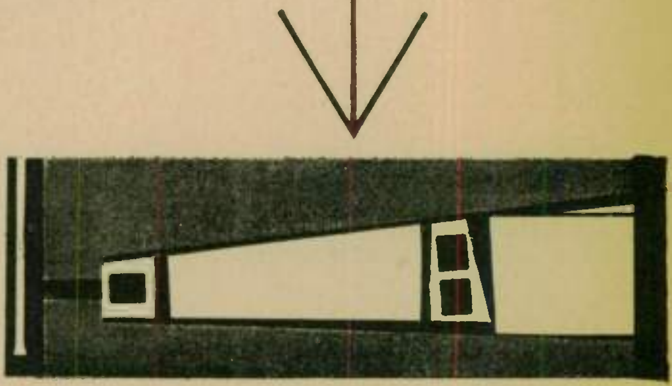
A ground bus, grounded at one point only on the chassis, is strongly recommended and the power supply should be constructed on a separate chassis. This will reduce the possibility of ground loops through the chassis and reduce hum to a minimum. The power supply should be capable of providing 300 volts at a minimum of 200 milliamperes. Under quiescent conditions total current drain is approximately 180 milliamperes. A center-tapped filament transformer (6.3v. @ 6a.) is recommended with the tap grounded. Either capacitor or choke input may be used.

Although the amplifier itself is perfectly stable, some instability at low frequencies may occur if the preamp is fed from a common voltage supply. If this occurs it may be necessary to use a separate supply.

Since construction of the first model of this amplifier our house has become a rendezvous for musicians, technicians, and at least one radio and TV program director, who come carrying their latest LP's for appraisal. One of them, a clarinet player, arrived on my doorstep first thing Christmas morning lugging six albums which had found their way down his chimney during the night. Critical listening tests over some months by this group have shown that the points outlined are well worth consideration.



Low-Noise Transistor Preamp



By **DAVID R. STEELE**
Ampex Corporation

Construction of simple, well-designed phono preamp for magnetic cartridges that can be built for under $15.

FOR the past decade, the serious record enthusiast has been plagued by problems involving equipment associated with magnetic cartridges. Foremost in this classification is the preamplifier. Commercial or homemade, they have been traditionally beset with microphonics, hum, and other noises that must be suppressed at every turn. This is a never-ending battle that has become more expensive and elaborate as the dynamic range of LP records has been steadily increased. It is now both simple and economical to overcome these problems with a low-noise transistorized preamplifier. Due to the nominal power requirement, it is feasible to use batteries, an excellent source of hum-free supply voltage. Transistors are non-microphonic, making mechanical precautions necessary. Transistors provide an economical approach to a low-noise, high-quality preamplifier.

It is universally accepted that it is the responsibility of the preamplifier to increase the output of the low-level magnetic cartridge and to provide correct equalization to complement the recording characteristic. These are both simple tasks. However, they must be performed without injecting any perceptible noise or distortion into the system. The preamplifier described herein accomplishes this at a total cost of less than $15, including batteries.

This is not intended to be a general-purpose unit. It is designed to do just one job and to do it as well as is technically possible. It should be evaluated on the same basis as any other preamplifier, namely noise, distortion, transient response, and accuracy of equalization. Naturally, the better these figures are, the better the preamplifier. But before any actual design work can be done, it is necessary to know exactly what this preamplifier is expected to do in terms of measurable performance.

These specifications represent desired performance. If subsequent investigation proves them extravagant or unattainable, they may be tempered with reality in the final specifications. Nevertheless, they provide a visible goal at the outset of the development and a yardstick by which its success may be measured upon completion. In this case, the following "wishing list" was compiled: Input impedance: 100,000 ohms; Output: 500 ohms or less; Gain: 35 db mid-frequency (to provide approximately 1 volt r.m.s. output); Distortion: less than 0.2 per-cent harmonic measured at maximum operating level +6 db; Equalization: RIAA ±2 db (RIAA tolerance); Noise: 80 db down from maximum operating level measuring all components in a band of 0 (d.c.) to 20 kc.

If this last specification seems unnecessarily detailed, it should be remembered that the signal-to-noise ratio is most informative when the frequency range of the measured noise is known. Unless otherwise specified, this might be taken only as the noise in the frequency range of interest, in this case, 30 cps to 20 kc. Unfortunately, noise components below 30 cps can be just as offensive to the rest of the system as those above. Therefore, it is felt that the noise measurement should be taken down to d.c.

Other desirable features include good transient response and fast overload recovery. Primarily, this goes to minimize the effect of dropping the stylus into the lead-in groove. It should be immune to stray fields generated by motors and transformers. The preamplifier should be compact and self-powered so that it may be installed close to the tone arm. It should operate reliably at temperatures up to 150°F. It must not use any special, selected, or unduly expensive components.

In consideration of the foregoing requirements, the design philosophy of a two-stage, direct-coupled feedback amplifier followed by a common collector stage, also direct-coupled, is very attractive. This configuration has definite advantages in that:

(*a*) Equalization can be accomplished in a single voltage feedback loop.

(*b*) The series feedback thus produced will elevate the input impedance.

(*c*) Temperature stabilization can be accomplished by a separate d.c. feedback loop.

(*d*) The common-collector output stage will provide a low-impedance output and isolate the equalizing network from the load.

Since the preamplifier is to be battery powered, it is practical to arrange the supply with two identical batteries to provide positive and negative voltage with respect to the mid-point, which is grounded, serving as a "0" voltage reference. The base of the input transistor is conveniently placed at ground potential due to the very low d.c. resistance of the cartridge. The base current of the first stage does flow through the cartridge, however, this is only about 2.5 microamperes and has no perceptible effect on the performance of the cartridge.

Circuit Employed

Referring to the schematic, Fig. 1, an absence of coupling capacitors will be noted. The direct coupling is responsible for the excellent overload recovery and low-frequency transient response. It is also responsible for the temperature stability, through the d.c. feedback loop between the input and output emitters. This method of temperature stabilization avoids the use of current-consuming divider networks at the base of each stage, resulting in considerably less battery drain than would otherwise be experienced.

For discussion purposes, the amplifier can be divided into three sections. The forward-gain path, the signal-feedback loop, and the d.c.-feedback loop. The forward-voltage gain is provided by V_1 and V_2, which are used in the common-emitter configuration. V_3, an emitter-follower, provides current gain, reducing the output impedance. A 2N-324, a high-*beta*, low-noise transistor, was selected for V_1. In order to obtain the desired d.c. distribution throughout the amplifier, it is necessary to use an *n-p-n* transistor for V_2. Type 2N169A was selected for this position. V_3, the output stage, is not at all critical. Another 2N324 is used here, reducing the number of different types required. These types are manufactured by *General Electric*. No difficulty should be encountered in the substitution of other types as long as they are electrically similar. Most manufacturers publish reliable cross-reference data, which can be used in the selection of other types. The parts list includes some alternate types that have been tried and found satisfactory. The use of an "economy" transistor in the V_1 position is not recommended, as this might degrade the noise figure of the preamplifier.

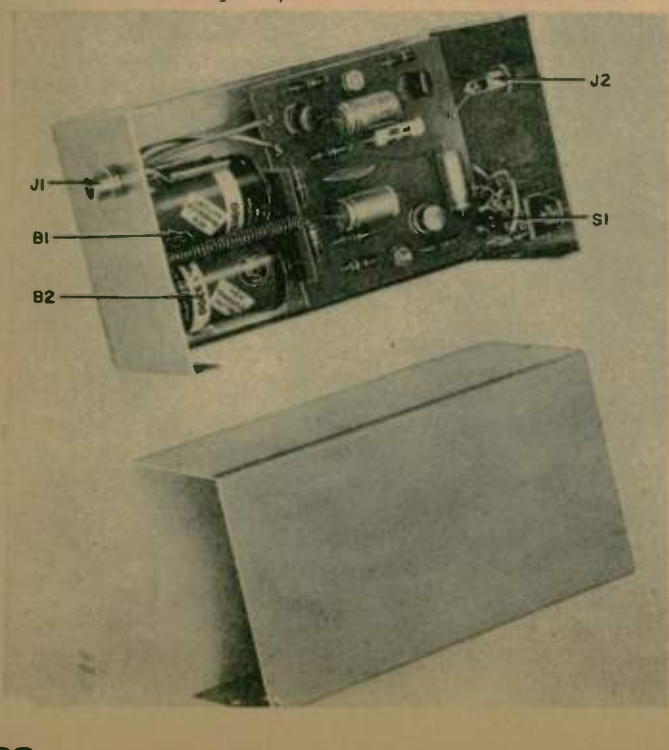
Signal feedback of a negative voltage characteristic is obtained from the collector of V_2, and developed across R_8, the unbypassed portion of the V_1 emitter resistance. This series feedback configuration raises the input impedance of the preamplifier to a value that prevents loading a high-impedance magnetic cartridge. Equalization is obtained through the use of a frequency-discriminating network in this loop. R_4, C_4, and C_5 are selected to produce the RIAA characteristic. R_4 and C_4 determine the low-frequency turnover point and C_5 determines the high-frequency roll-off. Naturally, the feed back is at its minimum at the top of the low-frequency rise. However, at this point there is still approximately 15 db of feedback through this loop.

The d.c. feedback loop includes the emitter-follower. It returns the d.c. component of the output to the input stage emitter. Any shift of d.c. operating points due to temperature, transistor characteristics, etc., would, without this over-all loop, produce a change in the d.c. voltage at the V_3 emitter. If this change is applied, in proper phase, to the input of the amplifier, the situation will be self-correcting, as it is in this case. The temperature stabilization realized in this manner is quite effective. Referring again to the schematic, it will be observed that C_1 is used to filter the signal out of the d.c. loop. This is necessary, since the signal feedback loop is "inside" the d.c. loop. Signal voltage appearing in the d.c. loop would tend to nullify the action of the equalizing network.

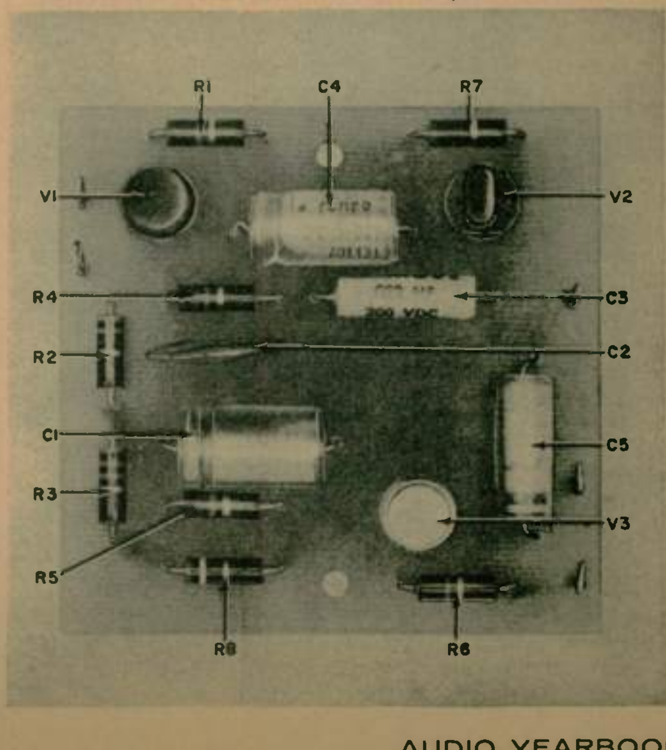
The capacity of C_1 is chosen so that, in conjunction with R_5, low-frequency feedback occurs in this loop. Actually C_1 and R_5 have been selected to permit feedback starting at 30 cps and increasing as frequency decreases. This serves to control the subaudible response of the preamplifier. It provides feedback in the region where the signal feedback loop has ceased to function. Three important benefits are derived from this feature. It is generally conceded that it is desirable to attenuate the response of the system below the range of interest to reduce turntable rumble and changer thumps. It reduces the possibility of low-frequency instability, and, most important, it greatly reduces the 1/F or flicker noise produced by transistors.

At first glance, the value of C_1 may

Inside view showing the placement of the two batteries used.



Circuit board on which transistors and other parts are mounted.



seem unusually large. To avoid confusion, it should be pointed out that R_5 is in series with an amplifier having an open loop gain of about 60 db, therefore, the effective resistance of R_5 is 27,000/1000 or 27 ohms. Nearly all LP recordings available today require the RIAA curve for correct reproduction and this is why the RIAA curve was built into this preamplifier. It is conceivable, however, that some users will desire other equalization, or perhaps several different switchable curves. Inasmuch as all of the equalization is accomplished by one network between two points, a single-pole rotary switch can be used to provide as many equalization positions as desired. If a curve having a 500 cps turnover and no high-end roll-off is desired, common to 78 rpm records, C_3 is changed to .02 µf. and C_2 is omitted. A capacitive reactance chart or nomograph is useful in determining the appropriate values to produce a particular curve. A general consideration that should be kept in mind is that the amount of equalization must always be less than the amount of feedback determined by R_4. Thus, if the turnover frequency is moved up, it should be done by decreasing the capacity of C_3 and decreasing the resistance of R_4, otherwise, the low-frequency rise may flatten out prematurely. By making C_3 very large (10 µf. or more) and omitting C_2, the response may be made essentially flat.

Input Impedance	120,000 ohms
Output Impedance	90 ohms
Gain	35 db @ 1000 cps
Distortion	0.17% @ 2 v. r.m.s. output (measured at 1 kc.)
Noise	−85 db, all components d.c. to 20 kc.
Equalization	RIAA
Output Level	Approx. 1 v. r.m.s.
Operating Temperature	165°F

Table 1. Listing of preamp specifications.

Performance specifications of the preamplifier are given in Table 1 and the response in Fig. 2. All of the original specifications have been met and, in some cases, exceeded. The only maintenance required—in fact the only maintenance possible—is the replacement of batteries when required. The exact battery life is, of course, dependent on how much the preamplifier is used. A good procedure is to check the voltage every three months or so, and discard the batteries when they measure less than approximately 6 volts.

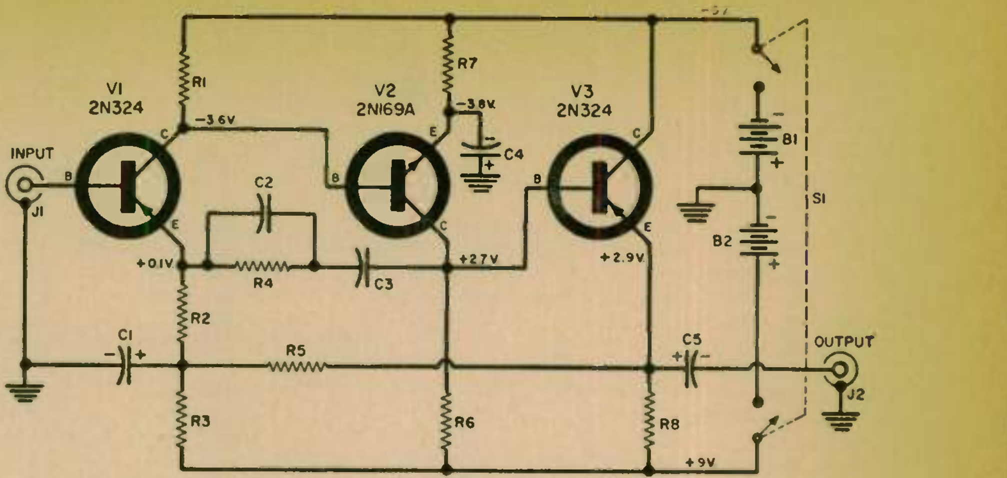
Layout & Construction

The physical layout and construction shown in the photographs are intended primarily as examples. Parts placement and lead dress are not at all critical. In the event the builder elects to forego sockets for the transistors, the usual precautions regarding soldering should be observed. The use of a d.p.s.t. relay connected to the turntable motor may be used to turn the preamplifier off and on. No doubt several possible innovations will occur to the reader. The "chassis" is 1/16" Fiberglas, but any of the common insulating materials will be satisfactory. The connecting terminals were made by bending a loop in the end of No. 20 tinned wire. The completed preamplifier is housed in a homemade aluminum box. It is recommended that this preamplifier be enclosed in some kind of a metallic shield.

During bench tests, the preamplifier input must be terminated by a low d.c. resistance. A 680-ohm resistor will be satisfactory. The d.c. voltages shown on the schematic were measured with a v.t.v.m. and also with a 100,000-ohm-per-volt multimeter. All readings should be ± 1 volt. The d.c. stability may be checked by connecting a voltmeter between V_3 emitter and ground. V_1 is then heated by holding a soldering iron in close proximity. The voltage will decrease. When it drops to about +1 volt, remove the soldering iron and grasp V_1 between thumb and forefinger. It should feel decidedly hot. The voltage at the V_3 emitter should return to normal rapidly as the heat is dissipated by the fingers.

Installation and Use

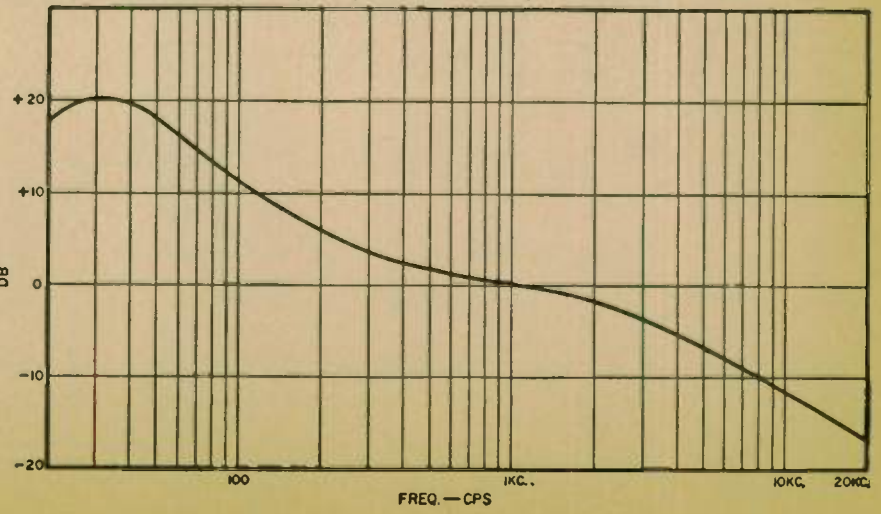
It is desirable to mount the preamplifier close to the turntable or changer, keeping the lead from the cartridge short. The higher impedance magnetic cartridges are sensitive to cable capacity. Since the preamplifier is not connected to the a.c. line, the possibility of ground loops in the system is slight. If encountered, the solution is merely to avoid connecting the preamplifier box to ground except by the shields of the input and output leads. It is preferable that the following amplifier have a volume control preceding the first stage, otherwise there is a possibility that the preamplifier will overdrive it, causing distortion. The simplest solution in this case is to insert a fixed attenuator between the preamplifier and the main amplifier.



R_1—15,000 ohm, ½ w. res.
R_2—330 ohm, ½ w. res.
R_3—33,000 ohm, ½ w. res.
R_4—18,000 ohm, ½ w. res.
R_5—27,000 ohm, ½ w. res.
R_6—10,000 ohm, ½ w. res.
R_7—8200 ohm, ½ w. res.
R_8—3900 ohm, ½ w. res.
C_1—250 µf., 3 v. elec. capacitor (see text)
C_2—.0039 µf. disc ceramic capacitor
C_3—.022 µf., 200 v. capacitor
C_4—100 µf., 6 v. elec. capacitor
C_5—1 µf., 6 v. capacitor
S_1—D.p.s.t. switch
B_1, B_2—9-volt battery (RCA VS300 or Burgess P6M)
J_1, J_2—Phono jack
V_1—"p-n-p" transistor (G-E 2N324, 2N508 or RCA 2N175)
V_2—"n-p-n" transistor (G-E 2N169A, RCA 2N585, Sylvania 2N214, or General Transistor 2N446A)
V_3—"p-n-p" transistor (G-E 2N324, 2N321, 2N508, 2N190, or RCA 2N109)

Fig. 1. Total cost of the three transistors used in this circuit runs about $6.50.

Fig. 2. Response curve of preamp shows how closely it follows RIAA characteristics.



$1\frac{7}{8}$-IPS

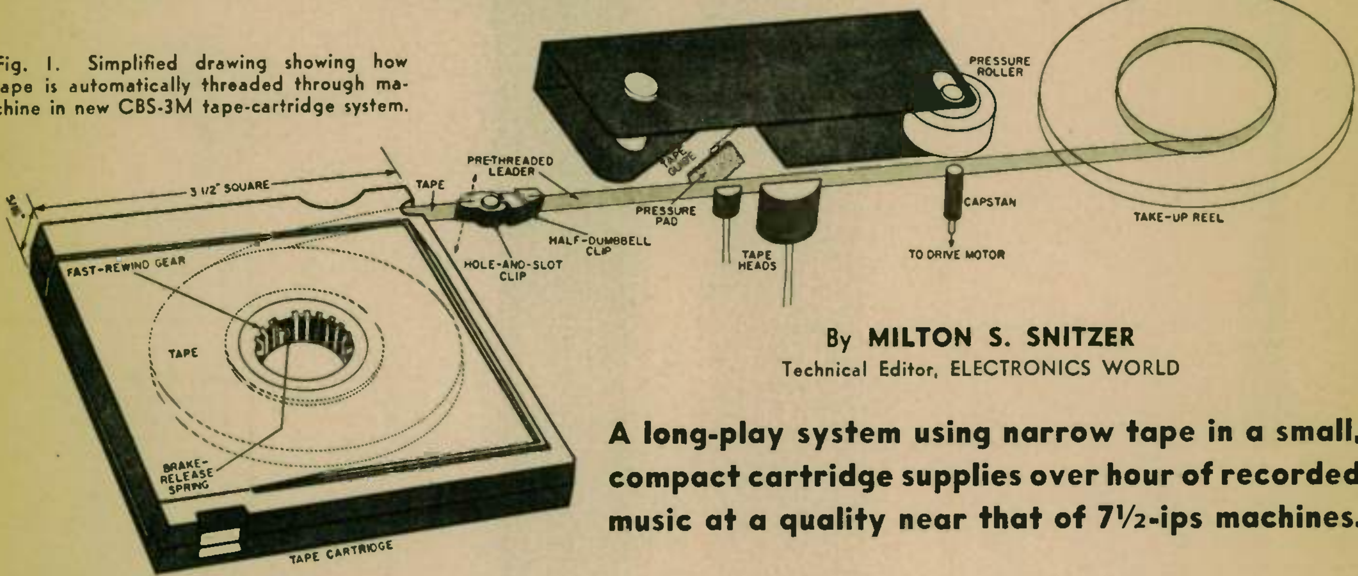


Fig. 1. Simplified drawing showing how tape is automatically threaded through machine in new CBS-3M tape-cartridge system.

By **MILTON S. SNITZER**
Technical Editor, ELECTRONICS WORLD

A long-play system using narrow tape in a small, compact cartridge supplies over hour of recorded music at a quality near that of 7½-ips machines.

Tape System for Stereo

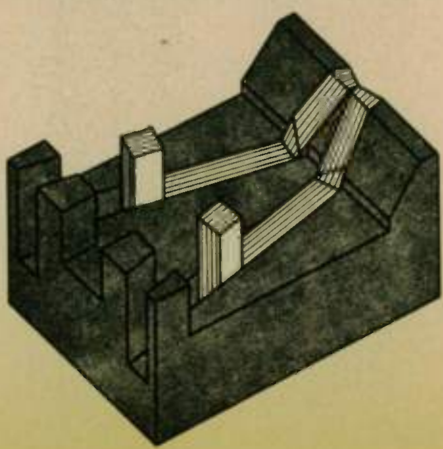
At a standing-room-only technical session during the recent IRE Convention, Dr. Peter C. Goldmark of *CBS Laboratories* demonstrated a new tape system developed for *Minnesota Minning and Manufacturing Co.* This new system, designed for home use, has been the subject of much speculation in the industry for the past few months. (For some details along these lines, refer to Bert Whyte's "Sound on Tape" in this issue.)

Heart of the tape player demonstrated is a small, compact tape cartridge a mere 3½ inches square and 5/16 inch thick. This cartridge is loaded with enough special narrow tape, only 150 mils wide (about ⅐ inch), to play continuously a stereo selection over an hour long. What is more, the player will accommodate five such tape cartridges which are played automatically in much the same way as an automatic record changer plays phonograph discs. Even with the ultra-slow speed of 1⅞ inches-per-second, performance is claimed to approach the best obtainable 7½-ips machines. The same instrument can also be used as a home recorder when loaded with blank-tape cartridges.

The demonstration conducted by *CBS* seemed to bear out the claims. At the conclusion of the technical papers a striking A-B test was conducted. One selection from "My Fair Lady" and a portion of the Mendelssohn "Violin Concerto" had been copied from the original 15-ips tape masters onto the new 1⅞-ips cartridge. This had then been copied onto a 15-ips half-track tape along with the same selections obtained directly from the master. The final tape was played back to the audience on a professional 15-ips tape machine. It was thus possible to make a direct comparison between the two systems.

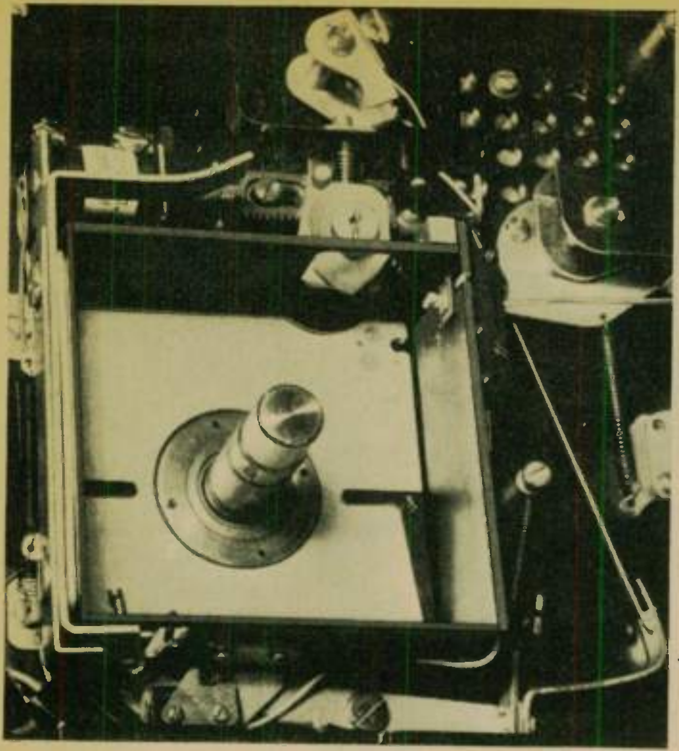
Although at least one of the selections was not particularly rich in extreme high frequencies, there was a remarkable similarity of sound when switching between the 1⅞-ips and the 15-ips versions. From where we were sitting in the first row of the large listening room, it was very easy to tell when switching was done but it was not at all easy, at least in some passages, to tell which version we were listening to. No wow or flutter could be heard in either version and good high-frequency response characterized both. *CBS* claimed an over-all flat frequency response from 30 to 15,000 cps, but this response is at the reduced level of −18 db relative to a level giving 3 per-cent distortion at 1 kc.

Fig. 2. Sub-assembly of playback head before coils are mounted on the laminations.



The tape machine used for the demonstration was an engineering prototype, and pre-production models of the unit are not expected until later this year. Because of the time required for tooling and actual production, no commercial instruments are expected for the home music market until 1961. Such instruments are to be marketed by *Zenith Radio Corp.* in the company's console models.

Immediately after the *CBS* disclosure and demonstration, *Ampex Corp.* announced that it too has a new recording technique that will make it possible to commercially record and duplicate tapes to play at 1⅞ ips. According to Herb Brown, *Ampex* vice-president, it will be at least another "two to three years" before pre-recorded tapes incorporating the new principles are developed to compete with the phonograph record in price and performance. He



Top view of prototype model of the new tape machine described. Stack of cartridges is pushed down into the square well, and button is depressed in order to begin the automatic operation.

Close-up of the tape well showing the large spindle on which the tape cartridges are loaded. Near the upper right-hand corner may be seen the half-dumbbell clip at end of the leader.

added that the new standard $7\frac{1}{2}$-ips, four-track, reel-to-reel tapes will remain a major factor in high quality home entertainment long after the slow-speed techniques are actively used in commercial production.

At the same time, Brown pointed out that if the industry can agree on a single standard of the basic magazine or cartridge concept, *Ampex* will cooperate by (1) disclosing details on its method of producing slow-speed recordings, (2) license its $1\frac{7}{8}$ recording process royalty-free, and (3) make available duplication equipment and conversion kits.

Brown said he foresees "harmonious co-existence" between the tape magazine and the $7\frac{1}{2}$-ips, four-track, reel-to-reel standard since each will appeal to a different segment of an ever-growing market.

Special Tape & Special Heads

One of the reasons for the success of the *CBS* system is the use of a new special magnetic tape, still only in pilot production at *3M*. This tape is 150 mils wide and 1 mil thick. Print-through does not appear to be much of a problem due to the shorter wavelengths that occur at the slow speeds and the special head design that is used. There is provision for three tracks on the tape, each 40 mils wide. Because of the narrow tracks used, head alignment becomes less of a problem. Two of the tracks are used for the conventional left and right stereo channels. The third track was suggested by Dr. Goldmark for use as a *delayed monophonic channel*. With this signal applied to a third amplifier and suitably spaced speaker, the important effect of concert-hall reverberation may be brought into the living room. The tape employed for the demonstration used only the usual two channels (left and right).

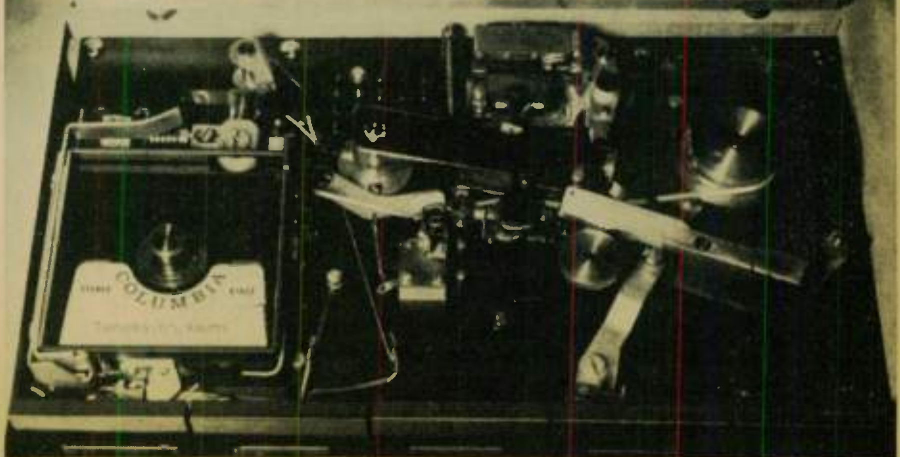
The new tape uses a special softer lacquer formulation that allows better head-to-tape contact with a minimum amount of particle rub-off. Also, smaller oxide particle size and better spread of particles result in less noise and improved high-frequency response.

The $3\frac{1}{2}$-inch-square cartridge contains enough of this tape to play continuously for 64 minutes so that it would be able to handle just about all classical music compositions without interruptions. The entire cartridge in its container occupies only about 4 cubic inches compared to a long-play phonograph record in its envelope which takes up about 20 cubic inches of storage space (12" x 12" x $\frac{1}{8}$").

Just as important in the new system is a specially designed playback head with the unbelievably narrow head gap of only 1 micron (about $\frac{1}{25}$th mil.) This gap is about $\frac{2}{5}$th the width of the narrowest gap in common use in present $3\frac{3}{4}$-ips, quarter-track tape machines. The construction of the head subassembly before the coils have been placed on the projecting laminations is shown in Fig. 2. This type of head can be constructed to produce an output of 1.5 mv. up to 15 kc. from a tape having a $\frac{1}{3}$-mil coating thickness.

When questioned about possible wear on a head having such a microscopically small gap, Dr. Goldmark replied that this was not much of a problem for the following reasons. Because of the low running speed of the tape and the softer, smoother tape surface, friction and heat are reduced. Also, a dust cover is used over the entire mechanism so that there is very little abrasive effect from dust particles.

Top view of prototype model of tape machine shown with the protective cover removed.



The record head that will be used with the *Zenith* consoles will probably record up to about 9 kc. However, a higher quality but more expensive record head could be made whose response was similar to the playback head.

Equalization and Performance

The normal maximum output curve of the replay head, when playing back a new tape at $1\frac{7}{8}$ ips made with a high efficiency record head, is shown in Fig. 3A. With the recording and playback equalization curves shown in Figs. 3B and 3C respectively, a flat response is obtained from 30 to 15,000 cps at −18 db with respect to a 1-kc., 3 per-cent distortion signal. Under these conditions the maximum signal-to-noise ratio at 1 kc. is 54 db, which is about the same obtainable on professional $7\frac{1}{2}$-ips half-track systems. These latter systems, however, are some 6 db better at a 10-kc. signal response than is the new $1\frac{7}{8}$-ips system.

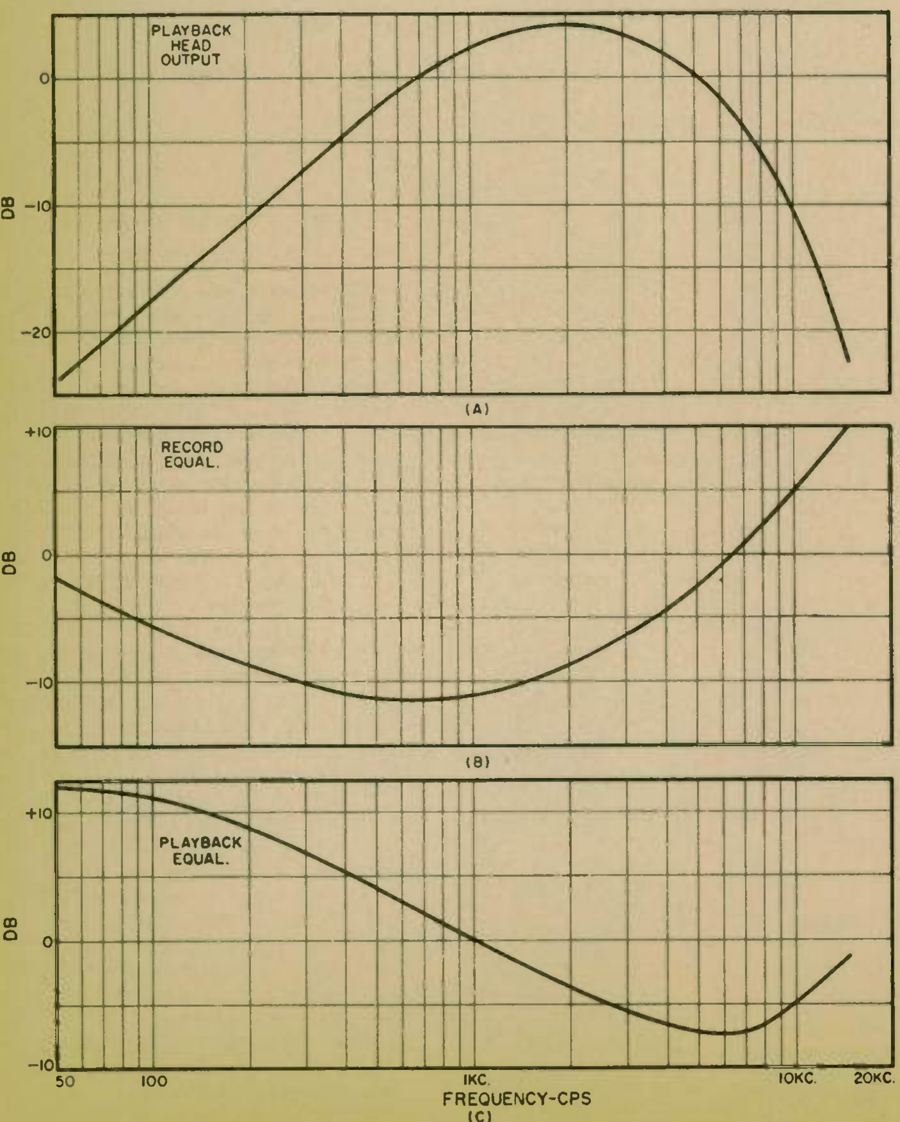
CBS therefore claims that this new system, taking into account the recording and playback equalizations used, approaches $7\frac{1}{2}$-ips performance closely and is entirely adequate for all types of musical selections.

Mechanical Features

One job that the layman or some home users find to be a nuisance is the loading and threading of a tape machine. In this new system, a number of tape cartridges are simply stacked on a wide spindle in a square well, the stack is pushed down to set the mechanism, and a button is depressed. (The spindle reminds us of the one used in some 45-rpm record changers.) The machine then proceeds to thread itself, the heads move into position, the take-up reel starts to wind in the tape, and the tape begins to play.

Then, at the end of a tape, or before this if the user presses the "reject" button, the machine stops and goes into a fast rewind (20 seconds for the entire reel). After the cartridge has been completely rewound, the entire stack of cartridges moves up by the thickness of one cartridge. The machine then proceeds to thread the tape from the new cartridge through the machine and play it back. How this is accomplished may be seen from Fig. 1 and some of the photographs of the uncovered mechanism.

Fig. 3. (A) Maximum output of playback head along with the recording equalization (B) and playback equalization (C) used. All curves apply to operation at $1\frac{7}{8}$-ips speed.



At the very beginning of the cartridge-loaded tape is a small clip with a hole and slot in it. This clip completely closes up the only small opening in the cartridge housing, thus protecting the tape. At the very end of the pre-threaded leader in the machine is another small clip with what looks like a half of a dumbbell protruding from it. Before the cartridges are loaded onto the machine, the dumbbell clip just extends inside the square well into which the cartridges are to be placed. When the tapes are put in place, the dumbbell clip just fits into the hole-and-slot clip, and the leader can now pull the tape out of the cartridge and through the machine to the take-up reel.

There is a straight-line path between the cartridge opening and the take-up reel so that it is not necessary to use intermediate idlers. A capstan and pressure roller draw the tape through the machine at the required speed of $1\frac{7}{8}$ ips. The take-up reel is driven through a conventional slipping clutch.

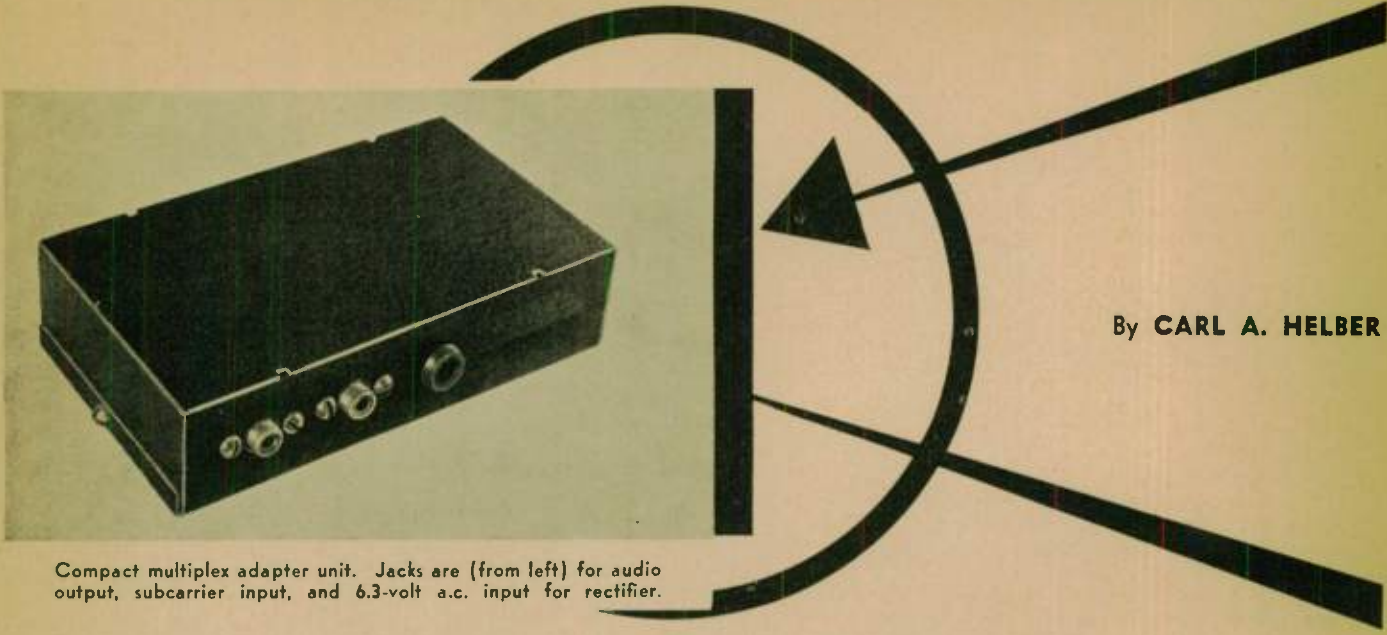
In order to prevent the tape from unreeling and spilling off the supply reel during normal handling, some sort of brake is required. This brake is built into the bottom wall of the cartridge in the form of ratchet teeth that are engaged by a simple spring clip. When the cartridge is loaded on the machine, the spindle presses against the spring clip, raises it from the ratchet teeth, and permits the supply reel to rotate freely within the cartridge. Also built into the cartridge hub is a toothed arrangement which acts as a fast-rewind gear. This is driven at high speed to rewind the tape quickly. The entire driving mechanism is carefully built so as to minimize wow and flutter, which is frequently found to be a serious problem at very low running speeds.

Costs and Future

We have no information on the possible costs of the tape cartridges or of the tape machine itself. Our guess is that the cartridge cost will be quite competitive with a stereo phonograph record, but the cost of the machine will probably not be low. Although the over-all mechanism is not much more complicated than a record changer, the tape machine requires more complex linkages from the drive motor and the design must be good mechanically to avoid wow and flutter at the slow tape speed used. Also, the heads themselves are not going to be cheap.

The disclosure and the demonstration by *CBS* certainly represents a notable technical achievement. What effect the introduction of this system is going to have on the tape industry and on consumers is something we will have to wait for the future to tell. And we don't have our crystal ball with us now.

Transistorized FM Multiplex Adapter



Compact multiplex adapter unit. Jacks are (from left) for audio output, subcarrier input, and 6.3-volt a.c. input for rectifier.

By **CARL A. HELBER**

Construction of experimental unit used with FM tuner to detect narrow-band multiplex signals. Requires separate matrixing circuits for multiplexed stereo.

HAVING recently completed the construction of an FM tuner kit, the author decided to design an experimental multiplex adapter which could be used with the tuner. Two excellent articles on the fundamentals of FM multiplex appeared in the January and February 1959 issues of this magazine and were used as the springboard for designing the adapter to be described.

The primary aim in designing this adapter was to gain some practical experience with FM multiplex systems. Several experimental devices were built in the course of this work and the one described is thought to be of interest to readers, from the standpoint of circuit simplicity.

While the unit in its present form is suitable only for narrow-band FM multiplex stereo, when used with suitable matrixing it can be modified for wide-band use. Considerably more tuner output is required for reliable operation, however.

To date the FCC has not decided which of the many proposed stereo multiplex systems to approve. In the interim the experimenter can content himself by becoming acquainted with the problems peculiar to some or all of the many types and can, if so inclined, accept the challenge of finding the simplest device capable of acceptable performance.

While previously published information applied to tube circuits, it appeared, after a cursory analysis, to be feasible to design an adapter using inexpensive germanium transistors. The advantages of such a device are simplicity, low power consumption, small size, and low cost. In fact the total cost of parts, including the 5½" x 3" x 1¼" case, was approximately $12.00.

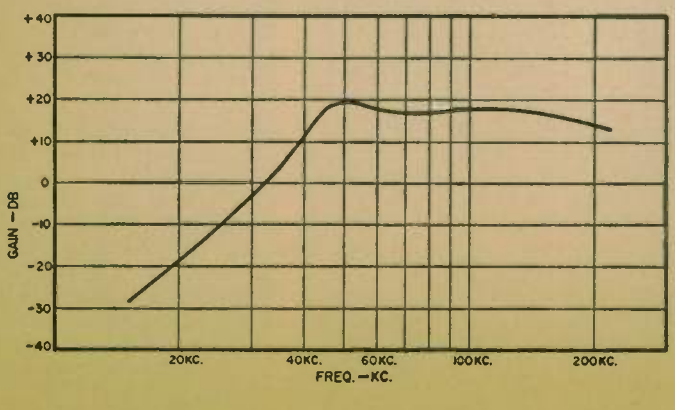


Fig. 1. Measured response of the filter and amplifier. Equipment was designed for a 67.5-kc. subcarrier so that the filter cuts off at around 40 kc. For a 50-kc. subcarrier, a cut-off frequency of about 25 kc. would be preferable.

Circuit Design

In the interests of simplicity a constant-*k*, high-pass filter and amplifier circuit was designed rather than the more complex bandpass type. The adapter described herein was for experiments on a subcarrier of 67.5 kc. and the filter was designed to cut-off in the neighborhood of 40 kc. For a 50 kc. subcarrier, a cut-off frequency of about 25 kc. would be more desirable. Fig. 4 is the diagram of the filter and amplifier while Fig. 1 shows the over-all gain characteristics as a function of frequency.

The filter was designed to match a 2700-ohm load as a reasonable compromise between the large physical size of the high-value inductors required for high-impedance filters and the excessive power required to drive a low-impedance filter to a given voltage output. A value of 2700 ohms is close enough to the input impedance of the transistor amplifier to provide a reasonable match for the filter. Measurements made on several 2N229 *n-p-n* transistors in the circuit shown gave an input impedance of about 3500 ohms at 67.5 kc. Mismatches of 2 to 1 showed no discernible ill effects as far as a hearing test was concerned.

The frequency discriminator was designed originally as a monostable multivibrator and is shown in Fig. 2. Transistor V_3 is normally held in saturation by base current flowing through R_5 while V_3 is conducting lightly because of its much lower base current. Transistor V_2 could be cut-off completely by decreasing the value of R_6 but at the expense of larger trigger requirements. When a positive pulse is applied to the base of V_2, the collector of V_2 drops sharply, as does the base of V_3. This action causes the collector of V_3 to rise and causes C_6 to start charging toward +6 volts. In about 4 or 5 microseconds C_6 has charged sufficiently, through R_8, to bring V_3 back into conduction, at which time the emitters of both transistors rise to a point where V_3 is conducting heavily again and V_2 only lightly.

Since the width of the generated pulse is essentially independent of the triggering rate, the average voltage appearing at the collector of V_3 will be

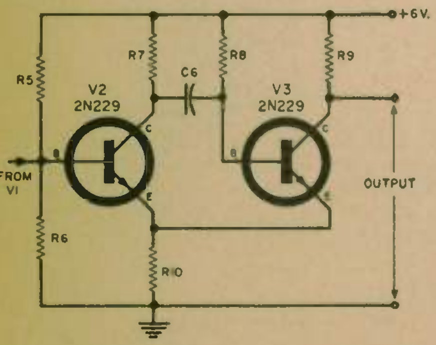


Fig. 2. Monostable multivibrator circuit.

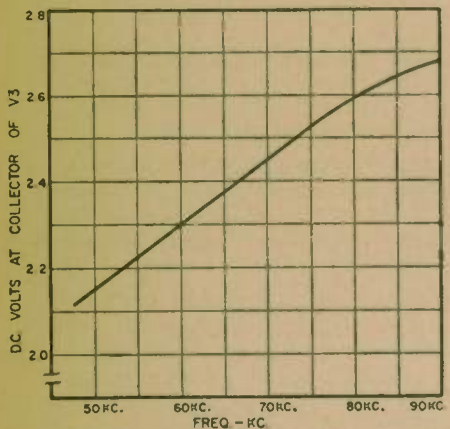


Fig 3. Monostable multivibrator response.

directly proportional to the triggering frequency up to the point where V_3 is off all the time. In this case the collector of V_3 would be at +6 volts. Usually before this condition is reached, the multivibrator begins to trigger on every other pulse and thus the average voltage at the collector of V_3 is linearly proportional to frequency over a limited range.

Fig. 3 shows the results of actual measurements taken on the circuit of Fig. 2 connected to the filter and amplifier of Fig. 4. A sinusoidal voltage of sufficient amplitude to cause reliable triggering of the multivibrator was applied to the input of the filter. From this curve it can be seen that a peak-to-peak audio component of about 0.22 volt will be developed for a maximum frequency deviation of ± 7.5 kc.

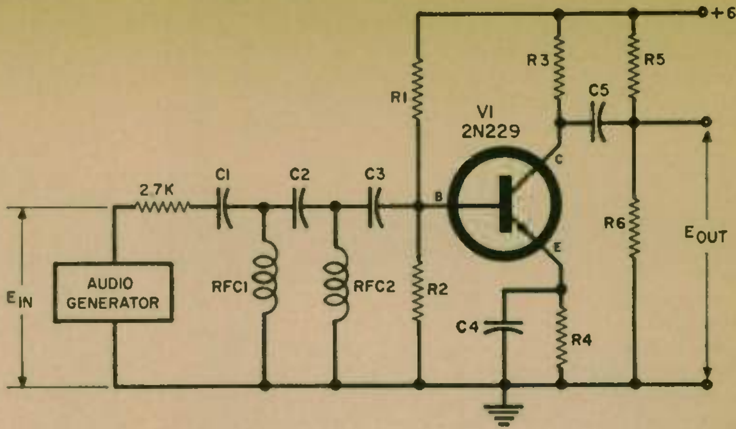


Fig. 4. Filter and amplifier portion of the transistorized multiplex adapter.

Measurements made on the overall circuitry showed that triggering started with an input signal of about 40 millivolts r.m.s. at 67.5 kc. and that no further change in d.c. level at the collector of V_3 occurred when the voltage increased beyond about 80 millivolts. Tests were made all the way up to 10 volts r.m.s. For this purpose a 2700-ohm resistor was connected between the filter input and the signal generator and the voltage measured at the generator end of the resistor.

In order to eliminate any possibility of distortion or overloading occurring between the FM tuner discriminator and the filter, it was decided to connect the filter directly to the tuner multiplex jack. It was necessary to replace the 100,000-ohm resistor normally present in the tuner with a 2700-ohm unit in order to achieve the necessary driving source impedance and high enough signal levels. No noticeable effect on normal FM reproduction appeared as a result of this change.

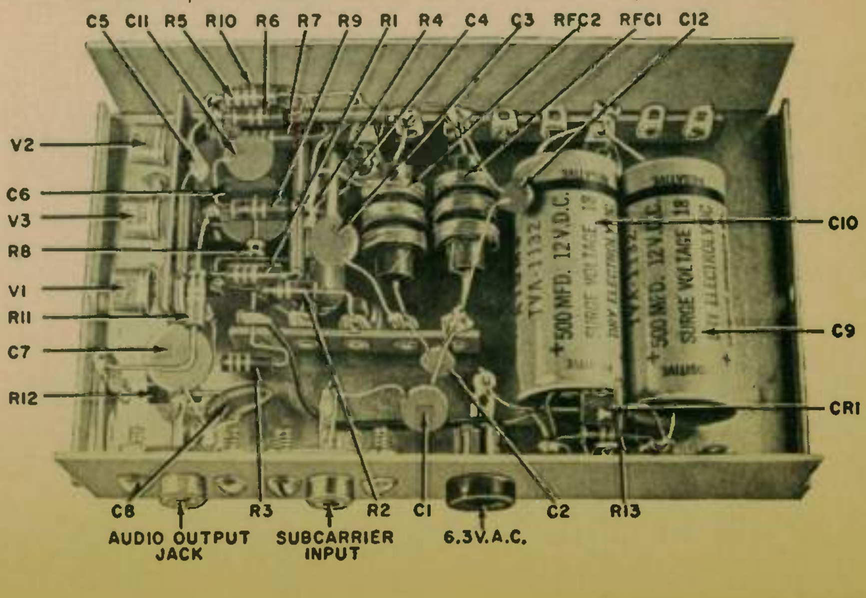
A filter between the collector of V_3 and the audio amplifier is required in order to keep the high-amplitude pulses from over-driving the input stage. The filter incorporated in the diagram provides adequate pulse filtering along with a substantial amount of de-emphasis of the high frequencies.

Power-supply measurements were made and showed that at 6 volts a current of 13 ma. was required. Since this was less than the current required by the average pilot light, it was decided to obtain power for the adapter from the rectified and filtered filament power of the tuner. The complete diagram of the unit, including the power supply, is shown in Fig. 5. Residual ripple is negligible.

Construction

Following the preliminary design, a circuit was built up and tests made. It was found experimentally that improved signal-to-noise ratio for initially noisy signals could be obtained by means of a slight modification of the filter (C_{12} was added to boost the subcarrier 4 or 5 db) and the monostable multivibrator was modified to a free-running multivibrator capable of being locked easily to the incoming modulated subcarrier. Nominally the multivibrator runs at about 65-70 kc. and

Bottom view of the adapter unit which is built inside a 5½" x 3" x 1¼" case. All the components have been labeled to minimize difficulty in construction.



puts out a somewhat triangular waveform. While for sufficient trigger levels the "off" time for V_3 remains fixed, the time between pulses is inversely proportional to the triggering frequency and thus the d.c. level at the collector of V_3 is proportional to frequency. The range of linear operation is limited but adequate for the ± 7.5 kc. subcarrier deviation systems.

One of the main reasons for improved signal-to-noise ratio with the free-running multivibrator is that should noise modulation of the subcarrier result in insufficient voltage to synchronize the multivibrator, the d.c. level would be somewhere in the mid-range of d.c. levels corresponding to the free-running multivibrator collector level. With the monostable multivibrator, insufficient trigger will cause the d.c. level to drop to less than a volt. For example, suppose that the average voltage to the collector of V_3 is 2 volts d.c. when the multivibrator is free-running or is 2 volts at, say, 70 kc. when the multivibrator is a monostable device with sufficient triggering amplitude. Suppose that at 67.5 kc. the d.c. level in both cases is 1.95 volts and that the d.c. level of the monostable m-v is 0.8 v. untriggered. If the free-running multivibrator fails to trigger properly the noise pulse produced is 2.00−1.95 or .05 volt peak, whereas for the monostable case it is 1.15 volts peak. In practice, because of a smaller area waveform, the free-running multivibrator produces a lower audio output (about .05 to 0.1 volt peak-to-peak) than the monostable multivibrator but is far more immune to noise in the subcarrier. Where adequate noise-free subcarrier signal is available, the monostable multivibrator gives more nearly linear operation over a wider frequency range.

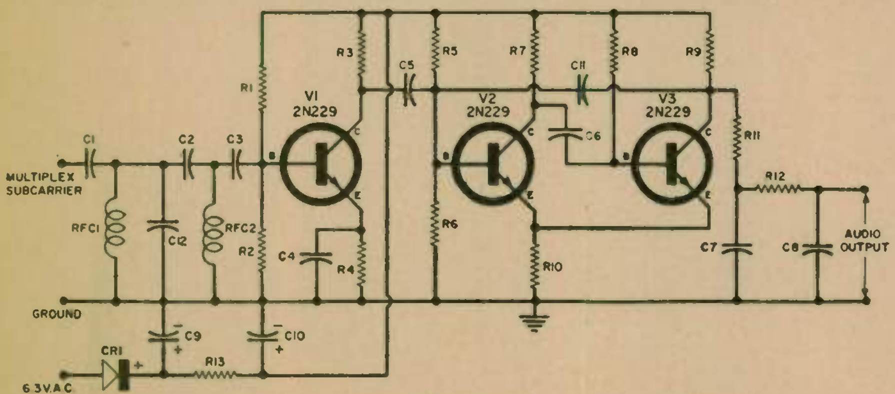
The accompanying photographs show one possible layout for the components. the layout is not at all critical but the circuitry should be completely enclosed to prevent pickup and demodulation of signals from powerful AM stations. The parts list accompanying the schematic diagram of Fig. 5 lists several specific parts but any equivalent components would be satisfactory. The circuit was designed to work with *Sylvania* 2N229 *n-p-n* transistors which are available for about 75 cents each, but will work equally well with similar *n-p-n* units of other manufacturers as well as *Sylvania's* 2N233's. By reversing the polarity of the power supply, many of the low-cost *p-n-p* units also worked very well.

Connections to the tuner are made by means of a length of shielded lead. A foot or two of lead is satisfactory. The connection to the ungrounded side of the heater in the tuner is made by means of a single flexible lead with *Amphenol* 71-1S plugs and 78-1S sockets installed in the tuner and adapter.

The tuner the author used employs a ratio detector but any tuner having a discriminator capable of supplying 100 to 200 millivolts r.m.s. of subcarrier into a 2700-ohm load should be satisfactory. For use with a 50-kc. subcarrier the filter constants should be changed to about 8 mhy. for RFC_1 and RFC_2, .0022 μf. for C_1 and C_3, and .001 μf. for C_2. The impedance of the filter will remain about 2700 ohms but the cut-off frequency will move down to about 25 kc. The values of C_6 and C_{11} should be changed to about 3900 $\mu\mu$f. and 3000 $\mu\mu$f. respectively.

As pointed out in earlier articles on FM multiplex, the use of an adapter to receive commercial transmissions for personal gain is a violation of Federal Law and makes the violator subject to criminal prosecution. There are no objections, however, to building an adapter such as this for personal and experimental purposes.

Fig. 5. Complete circuit diagram and parts listing for the multiplex adapter.



R_1—56,000 ohm, ½ w. carbon res.
R_3, R_{12}—10,000 ohm, ½ w. carbon res.
R_2—5600 ohm, ½ w. carbon res.
R_4, R_9—1000 ohm, ½ w. carbon res.
R_5—6800 ohm, ½ w. carbon res.
R_6—1500 ohm, ½ w. carbon res.
R_7—470 ohm, ½ w. carbon res.
R_8—3300 ohm, ½ w. carbon res.
R_{10}—68 ohm, ½ w. carbon res.
R_{11}—22,000 ohm, ½ w. carbon res.
R_{13}—100 ohm, ½ w. carbon res.
C_1, C_3—1500 $\mu\mu$f. ceramic capacitor (see text)
C_2—750 $\mu\mu$f. ceramic capacitor
C_4—0.1 μf., 200 v. paper capacitor
C_5—.005 μf. ceramic capacitor
C_6—.003 μf. ceramic capacitor (see text)
C_7, C_8—.01 μf. ceramic capacitor
C_9, C_{10}—500 μf., 12 v. elec. capacitor (Sprague TVA-1132)
C_{11}—.002 μf. ceramic capacitor (see text)
C_{12}—.001 μf. ceramic capacitor
RFC_1, RFC_2—5 mhy., 200 ma. r.f. choke (Miller 6304 or equiv.)
CR_1—1N91 or 1N92 germanium diode
V_1, V_2, V_3—"n-p-n" transistor (Author used Sylvania 2N229's. See text for alternates)

Multiplexing Music With One Recorder

By **ROBERT H. SHAW**

Simple modification of tape recorder allows one to sing a duet, trio, or accompany oneself with 2-3 instruments.

NOW you can sing a duet, trio, or perhaps accompany yourself with as many as three other instruments. Here's how your present recorder can be adapted to make "multiplexed" recordings at relatively low cost. If your machine already has provision for stereo playback, half the battle is over. The method to be described solves the problem of maintaining "balance" between the different parts and of timing.

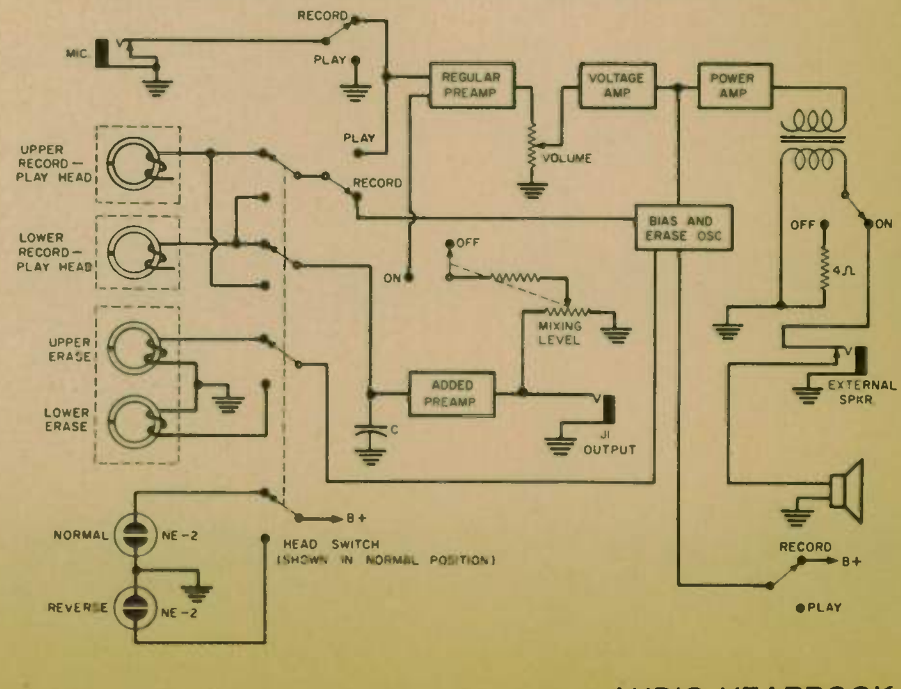
With a little experience in level setting, you will be able to turn out quality multiple recordings that will delight you and your friends.

How It Works

Here, briefly, is how the idea works: Referring to Fig. 1, we divide the tape into two parts—upper and lower tracks—as is done in regular stereo recordings. Then, using the "Head Switch," we alternate the recording head from upper to lower track at the same time the playback head is switched. After recording the first part on the upper track, we switch heads so that the upper track is now played back and, by means of a "Mixing Level" control, it is added to the lower track at the same time as you add the second part. The heads are alternated again and the procedure repeated until the desired number of parts are recorded. By listening through headphones plugged into the "External Speaker" jack, you can tell quite accurately if you are recording the second part too loud or too soft and can regulate yourself accordingly.

Excellent results were obtained with three-part recordings, but four or five parts cut down the quality of the first part. The recorder the author modified was a *Voice of Music* Model 710. This unit had no provision for stereo playback so the preamp circuits of Fig. 2 and two new stereo record-playback heads were added. This is similar to the circuit used in the regular stereo playback system of the *VM* 711, with modifications to compensate for shunt capacitance across the input. The stereo record-playback heads used were Type B, manufactured by *Michigan Magnetics Inc.* of Vermontville, Michigan. The erase heads were Type SSB from the same company. These were physically interchangeable with the original *VM* heads without any alterations in mounting.

Fig. 1. Block diagram of the modified tape recorder employed by the author.



If your recorder has provision for stereo playback then you will only have to add the extra erase head for the lower channel, the head switch, and the mixing level control. It will also be necessary to add sufficient capacity across your stereo preamp input to keep the bias oscillator from saturating it while recording. The coupling between the oscillator and preamp takes place in the recording heads. If you find this shunt capacitor noticeably affects the "highs," then change the value of your equalizing component to compensate for it. Don't let this discourage you, as it often involves changing a single component. Fig. 3 indicates the possible components making up your particular equalizing circuit. You can still have regular stereo operation by merely turning the "Mixing Level" control off and adding an amplifier for the second channel.

Procedure to be Followed

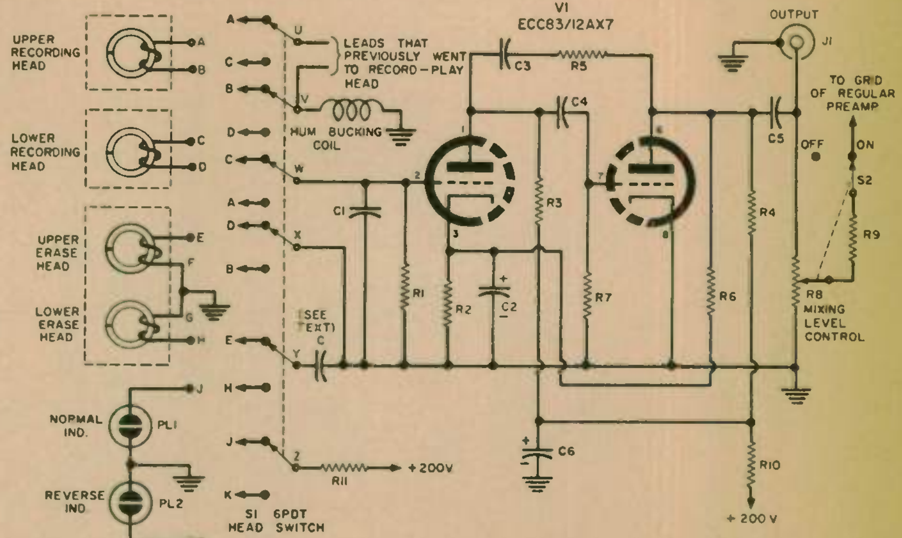
Although the following procedure is given for the *VM* Model 710 recorder, it will work equally well with most home recorders. These steps are given in the proper sequence for construction. *Important! Make certain the tools you use around the recording heads are not magnetized!*

First install the stereo record-playback heads and erase heads, lining them up as shown in Fig. 4. Mount the head indicator neons. The author merely drilled holes in the plastic head cover and cemented an NE-2 neon in each hole. These neons will probably last as long as the recorder and their use conveniently eliminates the need for lamp sockets, lens, etc. Don't let them protrude too far through the hole as they could get broken off.

Next mount the head switch, phono jack, and mixing level control in an accessible location, keeping them away from the electrical fields of the power transformer and motor. Now find a spot to mount the preamp, preferably near the original preamp and as far away from the power supply circuit as possible. Cut the required holes and put in grommets. Next, solder thin copper strips to the tube socket to serve as a shock-mount for the preamp, thus reducing the chances of microphonics. Mount the tube socket and then wire in the filaments, being careful to dress the leads away from pins 2 and 7.

Now label the head switch as shown in Fig. 2. This makes identification and wiring much simpler. Now wire all common points together, *i.e.*, "A" to "A," "B" to "B," etc. Next take the leads that went to your old record-play head and re-route them to points "U" and "V" of the head switch. Wire in the newly installed record-play heads according to details shown in Fig. 2. Remember to ground only one end of the shield. One of the leads that went to the erase is grounded. Take the other "hot" lead and re-route it to point "Y" of the switch. Wire up points "E" and "H" to the same "E" and "H" points on the head switch. Likewise wire in the neon indicators to points "J" and "K" of the switch. Wire in the voltage to light the neons to point "Z," adding the 1-megohm resistor in series.

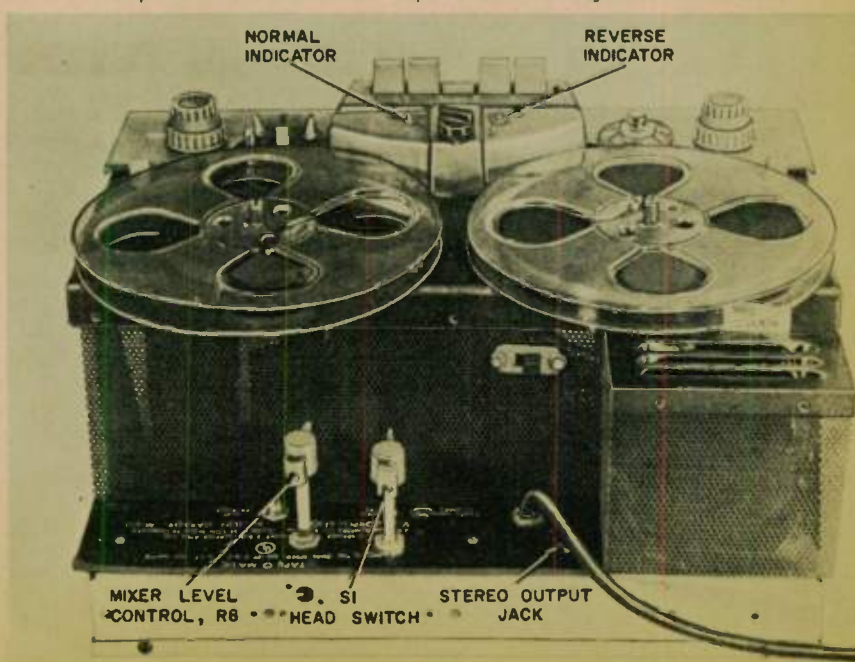
Wire in the other components, keeping all leads as short and direct as possible and using shielded cable for longer leads. Regular plastic-covered shielded microphone cable was used in the author's unit with very good results. The output of your new preamp, coming off the .047 µf. (C_5) capacitor, is to be wired to either the first or second amplifier stage of your regular recorder circuit. The author used the second stage, pin 7 of the normal preamp, as it provided sufficient gain. If more gain is required, you can use the first stage and change R_9 from 470,000 ohms to 1 megohm to minimize interaction between the "Mixing Level" control, R_8, and the microphone level.



R_1, R_6, R_{11}—1 megohm, ½ w. res.
R_2—4700 ohm, ½ w. res.
R_3, R_5—220,000 ohm, ½ w. res.
R_4, R_{10}—100,000 ohm, ½ w. res.
R_7—10 megohm, ½ w. res.
R_8—500,000 ohm audio-taper pot (with switch, S_2)
R_9—470,000 ohm, ½ w. res. (see text)
C_1, C_4—.01 µf., 400 v. capacitor
C_2—25 µf., 25 v. elec. capacitor
C_3—135 µµf. ceramic capacitor
C_5—.047 µf., 400 v. capacitor
C_6—20 µf., 400 v. elec. capacitor
S_1—6-pole, d.t. switch (Centralab 2018 or equiv.)
S_2—S.p.s.t. switch (on R_8)
PL_1, PL_2—NE-2 neon bulb (NE-51 if bayonet lamp socket is used)
J_1—Phono jack
2—Stacked stereo record/playback head (Michigan Magnetics Type B or equiv. See text)
2—Matched erase heads (Michigan Magnetics Type SSB or equiv.)
V_1—ECC83/12AX7 tube

Fig. 2. Complete schematic diagram of the circuit modifications made.

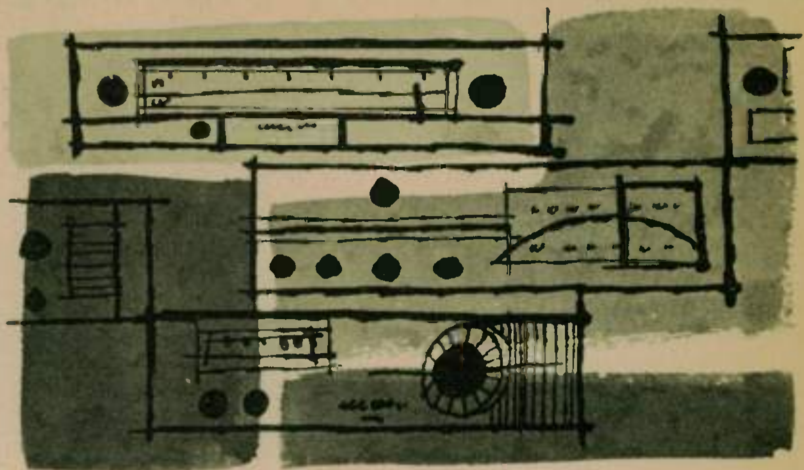
Top rear view of the modified tape recorder showing the added controls.



the fun they are missing.

only a few dollars.

new fm am tuner standard



By **DANIEL R. VON RECKLINGHAUSEN**
Chairman, IHFM Sub-Committee on Tuners
Chief Research Engineer, H. H. Scott, Inc.

Salient features of the Hi-Fi Institute's recently adopted standards for measuring AM/FM tuners.

THE Institute of High Fidelity Manufacturers has recently released its first standard, IHFM-T-100, entitled "Methods of Measurement for Tuners."

This standard represents a step forward in the techniques of tuner measurement. Previously, the only standards covering this field were those released by the Institute of Radio Engineers in 1947-48. These IRE standards were the best that could be devised at the time but later experience showed that certain necessary characteristics of tuner performance had not been included while other characteristics were measured in such a way that "good" test results did not necessarily mean "good" listening performance.

According to this new standard, a number of characteristics of either or both AM and FM tuners are to be measured: tuning range and frequency calibration, usable sensitivity, volume sensitivity, capture ratio, selectivity, amplitude modulation suppression, frequency response, distortion, spurious response, hum and noise, frequency drift, radiation, automatic frequency control, squelch control, loop antenna, and crosstalk.

Standard Conditions

All of these listed characteristics are to be measured under "standard" conditions. These conditions are:

Carrier frequencies: For FM tests, 90, 98, and 106 mc. and for AM tests 600, 1000, and 1400 kc. are standard test frequencies. Some tests are to be made using only one carrier frequency in which case it will be 98 mc. for FM and 1000 kilocycles for an AM tuner.

Since the settings of the various tuner controls will affect test results, certain standardized control positions have been specified. For example, the a.f.c. control, if present, is set for minimum control; the squelch control for maximum sensitivity; any tone controls for flattest response as indicated by panel markings; and volume control for maximum. Only in the case of tuners which incorporate extra stages of amplification, such as units equipped with phono-preamplifier or power-output stages, should the volume control be set for 20 db attenuation. This is to prevent clipping of the audio signal due to excessive amplification.

Input voltages, their measurement, and dummy antennas to be used for both AM and FM testing have been specified. These standards are diagrammed in Figs. 1 through 4.

The standard test modulation has also been specified. For most tests, the modulation frequency, except for some distortion and frequency response tests, is 400 cps. The standard modulation for FM is 100% or 75 kc. deviation while for AM this figure is 30%, except for distortion tests, where 90% is to be used.

Usable Sensitivity

If one wished to specify the most important characteristic of an FM tuner, a measurement of "usable sensitivity" would have to be made. The definition and measurement of usable sensitivity constitute the most important contribution made in the new standard. This term will be used generally within a few months to describe the sensitivity of tuners rather than the "quieting" figure as defined in the IRE standards and used up to now. The best definition and test method covering usable sensitivity of FM tuners is found in Section 6.03.02 of the new standard. Here is this section in its entirety:

"This test is performed at each of the standard test frequencies with the signal generator connected to the tuner under test through the standard 300-ohm dummy antenna. The signal generator should be frequency modulated with standard test modulation. The controls of the tuner shall be set to the normal control settings. The signal intensity should then be reduced to the least value which will produce a 30 decibel rise in indicated output with standard test modulation as compared with the indicated output with standard test modulation measured through a 400-cps null filter. This test serves to indicate the relative freedom of the tuner from objectionable internal receiver noise during pauses in modulation and receiver noise is least likely to be masked by modulation. This test also serves to indicate the relative freedom of the tuner from objectionable distortion during periods of maximum modulation.

"The results are expressed in microvolts."

What is new and different about this definition and how does it help the user to decide about the performance of a tuner?

First, this and all other measurements in the new standard are to be

If your recorder has provision for stereo playback then you will only have to add the extra erase head for the lower channel, the head switch, and the mixing level control. It will also be necessary to add sufficient capacity across your stereo preamp input to keep the bias oscillator from saturating it while recording. The coupling between the oscillator and preamp takes place in the recording heads. If you find this shunt capacitor noticeably affects the "highs," then change the value of your equalizing component to compensate for it. Don't let this discourage you, as it often involves changing a single component. Fig. 3 indicates the possible components making up your particular equalizing circuit. You can still have regular stereo operation by merely turning the "Mixing Level" control off and adding an amplifier for the second channel.

Procedure to be Followed

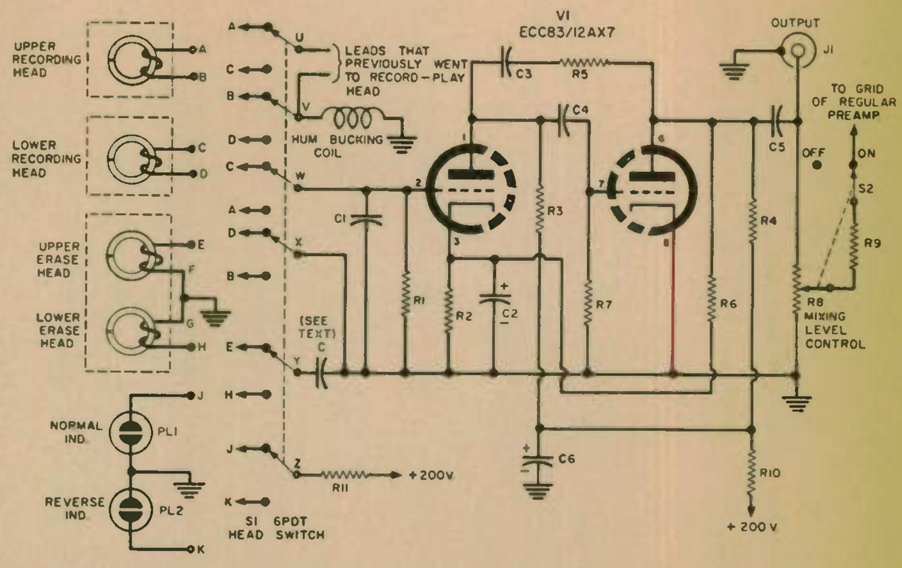
Although the following procedure is given for the *VM* Model 710 recorder, it will work equally well with most home recorders. These steps are given in the proper sequence for construction. *Important! Make certain the tools you use around the recording heads are not magnetized!*

First install the stereo record-playback heads and erase heads, lining them up as shown in Fig. 4. Mount the head indicator neons. The author merely drilled holes in the plastic head cover and cemented an NE-2 neon in each hole. These neons will probably last as long as the recorder and their use conveniently eliminates the need for lamp sockets, lens, etc. Don't let them protrude too far through the hole as they could get broken off.

Next mount the head switch, phono jack, and mixing level control in an accessible location, keeping them away from the electrical fields of the power transformer and motor. Now find a spot to mount the preamp, preferably near the original preamp and as far away from the power supply circuit as possible. Cut the required holes and put in grommets. Next, solder thin copper strips to the tube socket to serve as a shock-mount for the preamp, thus reducing the chances of microphonics. Mount the tube socket and then wire in the filaments, being careful to dress the leads away from pins 2 and 7.

Now label the head switch as shown in Fig. 2. This makes identification and wiring much simpler. Now wire all common points together, *i.e.*, "A" to "A," "B" to "B," etc. Next take the leads that went to your old record-play head and re-route them to points "U" and "V" of the head switch. Wire in the newly installed record-play heads according to details shown in Fig. 2. Remember to ground only one end of the shield. One of the leads that went to the erase is grounded. Take the other "hot" lead and re-route it to point "Y" of the switch. Wire up points "E" and "H" to the same "E" and "H" points on the head switch. Likewise wire in the neon indicators to points "J" and "K" of the switch. Wire in the voltage to light the neons to point "Z," adding the 1-megohm resistor in series.

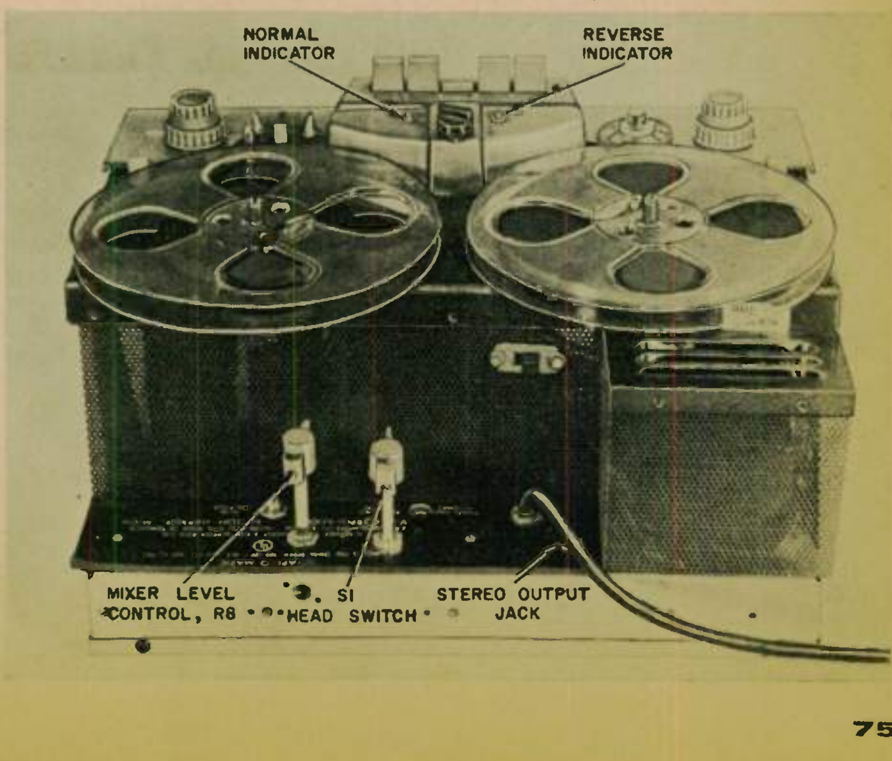
Wire in the other components, keeping all leads as short and direct as possible and using shielded cable for longer leads. Regular plastic-covered shielded microphone cable was used in the author's unit with very good results. The output of your new preamp, coming off the .047 μf. (C_5) capacitor, is to be wired to either the first or second amplifier stage of your regular recorder circuit. The author used the second stage, pin 7 of the normal preamp, as it provided sufficient gain. If more gain is required, you can use the first stage and change R_9 from 470,000 ohms to 1 megohm to minimize interaction between the "Mixing Level" control, R_8, and the microphone level.



R_1, R_6, R_{11}—1 megohm, ½ w. res.
R_2—4700 ohm, ½ w. res.
R_3, R_4—220,000 ohm, ½ w. res.
R_5, R_{10}—100,000 ohm, ½ w. res.
R_7—10 megohm, ½ w. res.
R_8—500,000 ohm audio-taper pot (with switch, S_2)
R_9—470,000 ohm, ½ w. res. (see text)
C_1, C_4—.01 μf., 400 v. capacitor
C_2—25 μf., 25 v. elec. capacitor
C_3—135 $\mu\mu$f. ceramic capacitor
C_5—.047 μf., 400 v. capacitor
C_6—20 μf., 400 v. elec. capacitor
S_1—6-pole, d.t. switch (Centralab 2018 or equiv.)
S_2—S.p.s.t. switch (on R_8)
PL_1, PL_2—NE-2 neon bulb (NE-51 if bayonet lamp socket is used)
J_1—Phono jack
2—Stacked stereo record/playback head (Michigan Magnetics Type B or equiv. See text)
2—Matched erase heads (Michigan Magnetics Type SSB or equiv.)
V_1—ECC83/12AX7 tube

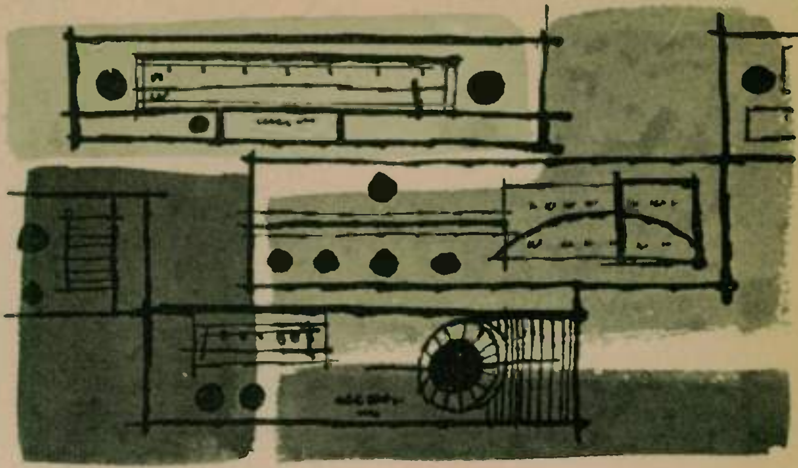
Fig. 2. Complete schematic diagram of the circuit modifications made.

Top rear view of the modified tape recorder showing the added controls.



1961 EDITION
77

new fm am tuner standard



By **DANIEL R. VON RECKLINGHAUSEN**

Chairman, IHFM Sub-Committee on Tuners
Chief Research Engineer, H. H. Scott, Inc.

Salient features of the Hi-Fi Institute's recently adopted standards for measuring AM/FM tuners.

THE Institute of High Fidelity Manufacturers has recently released its first standard, IHFM-T-100, entitled "Methods of Measurement for Tuners."

This standard represents a step forward in the techniques of tuner measurement. Previously, the only standards covering this field were those released by the Institute of Radio Engineers in 1947-48. These IRE standards were the best that could be devised at the time but later experience showed that certain necessary characteristics of tuner performance had not been included while other characteristics were measured in such a way that "good" test results did not necessarily mean "good" listening performance.

According to this new standard, a number of characteristics of either or both AM and FM tuners are to be measured: tuning range and frequency calibration, usable sensitivity, volume sensitivity, capture ratio, selectivity, amplitude modulation suppression, frequency response, distortion, spurious response, hum and noise, frequency drift, radiation, automatic frequency control, squelch control, loop antenna, and crosstalk.

Standard Conditions

All of these listed characteristics are to be measured under "standard" conditions. These conditions are:

Carrier frequencies: For FM tests, 90, 98, and 106 mc. and for AM tests 600, 1000, and 1400 kc. are standard test frequencies. Some tests are to be made using only one carrier frequency in which case it will be 98 mc. for FM and 1000 kilocycles for an AM tuner.

Since the settings of the various tuner controls will affect test results, certain standardized control positions have been specified. For example, the a.f.c. control, if present, is set for minimum control; the squelch control for maximum sensitivity; any tone controls for flattest response as indicated by panel markings; and volume control for maximum. Only in the case of tuners which incorporate extra stages of amplification, such as units equipped with phono-preamplifier or power-output stages, should the volume control be set for 20 db attenuation. This is to prevent clipping of the audio signal due to excessive amplification.

Input voltages, their measurement, and dummy antennas to be used for both AM and FM testing have been specified. These standards are diagrammed in Figs. 1 through 4.

The standard test modulation has also been specified. For most tests, the modulation frequency, except for some distortion and frequency response tests, is 400 cps. The standard modulation for FM is 100% or 75 kc. deviation while for AM this figure is 30%, except for distortion tests, where 90% is to be used.

Usable Sensitivity

If one wished to specify the most important characteristic of an FM tuner, a measurement of "usable sensitivity" would have to be made. The definition and measurement of usable sensitivity constitute the most important contribution made in the new standard. This term will be used generally within a few months to describe the sensitivity of tuners rather than the "quieting" figure as defined in the IRE standards and used up to now. The best definition and test method covering usable sensitivity of FM tuners is found in Section 6.03.02 of the new standard. Here is this section in its entirety:

"This test is performed at each of the standard test frequencies with the signal generator connected to the tuner under test through the standard 300-ohm dummy antenna. The signal generator should be frequency modulated with standard test modulation. The controls of the tuner shall be set to the normal control settings. The signal intensity should then be reduced to the least value which will produce a 30 decibel rise in indicated output with standard test modulation as compared with the indicated output with standard test modulation measured through a 400-cps null filter. This test serves to indicate the relative freedom of the tuner from objectionable internal receiver noise during pauses in modulation and receiver noise is least likely to be masked by modulation. This test also serves to indicate the relative freedom of the tuner from objectionable distortion during periods of maximum modulation.

"The results are expressed in microvolts."

What is new and different about this definition and how does it help the user to decide about the performance of a tuner?

First, this and all other measurements in the new standard are to be

If your recorder has provision for stereo playback then you will only have to add the extra erase head for the lower channel, the head switch, and the mixing level control. It will also be necessary to add sufficient capacity across your stereo preamp input to keep the bias oscillator from saturating it while recording. The coupling between the oscillator and preamp takes place in the recording heads. If you find this shunt capacitor noticeably affects the "highs," then change the value of your equalizing component to compensate for it. Don't let this discourage you, as it often involves changing a single component. Fig. 3 indicates the possible components making up your particular equalizing circuit. You can still have regular stereo operation by merely turning the "Mixing Level" control off and adding an amplifier for the second channel.

Procedure to be Followed

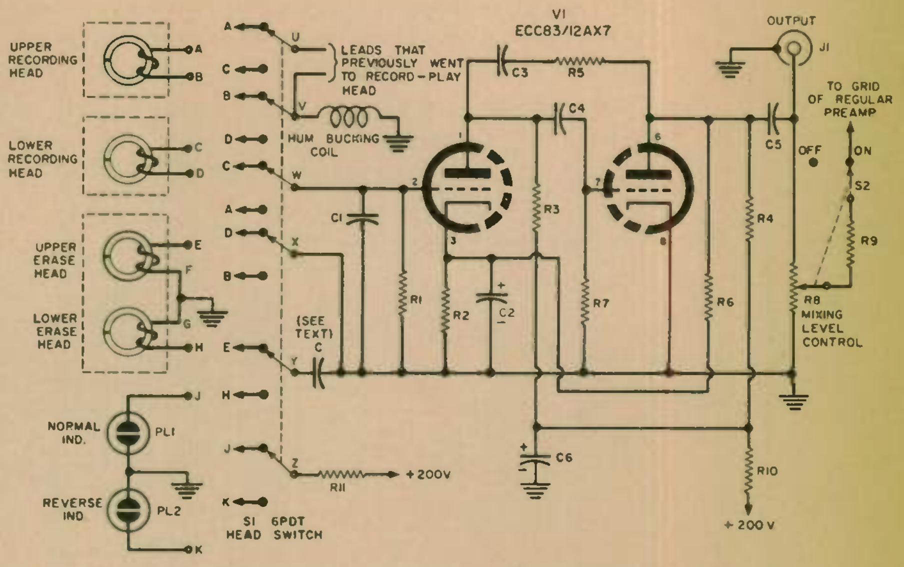
Although the following procedure is given for the *VM* Model 710 recorder, it will work equally well with most home recorders. These steps are given in the proper sequence for construction. *Important! Make certain the tools you use around the recording heads are not magnetized!*

First install the stereo record-playback heads and erase heads, lining them up as shown in Fig. 4. Mount the head indicator neons. The author merely drilled holes in the plastic head cover and cemented an NE-2 neon in each hole. These neons will probably last as long as the recorder and their use conveniently eliminates the need for lamp sockets, lens, etc. Don't let them protrude too far through the hole as they could get broken off.

Next mount the head switch, phono jack, and mixing level control in an accessible location, keeping them away from the electrical fields of the power transformer and motor. Now find a spot to mount the preamp, preferably near the original preamp and as far away from the power supply circuit as possible. Cut the required holes and put in grommets. Next, solder thin copper strips to the tube socket to serve as a shock-mount for the preamp, thus reducing the chances of microphonics. Mount the tube socket and then wire in the filaments, being careful to dress the leads away from pins 2 and 7.

Now label the head switch as shown in Fig. 2. This makes identification and wiring much simpler. Now wire all common points together, *i.e.*, "A" to "A," "B" to "B," etc. Next take the leads that went to your old record-play head and re-route them to points "U" and "V" of the head switch. Wire in the newly installed record-play heads according to details shown in Fig. 2. Remember to ground only one end of the shield. One of the leads that went to the erase is grounded. Take the other "hot" lead and re-route it to point "Y" of the switch. Wire up points "E" and "H" to the same "E" and "H" points on the head switch. Likewise wire in the neon indicators to points "J" and "K" of the switch. Wire in the voltage to light the neons to point "Z," adding the 1-megohm resistor in series.

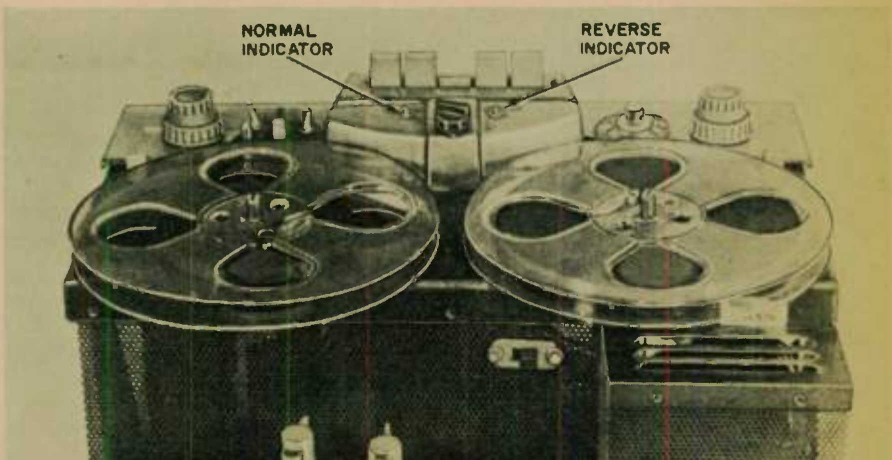
Wire in the other components, keeping all leads as short and direct as possible and using shielded cable for longer leads. Regular plastic-covered shielded microphone cable was used in the author's unit with very good results. The output of your new preamp, coming off the .047 μf. (C_5) capacitor, is to be wired to either the first or second amplifier stage of your regular recorder circuit. The author used the second stage, pin 7 of the normal preamp, as it provided sufficient gain. If more gain is required, you can use the first stage and change R_9 from 470,000 ohms to 1 megohm to minimize interaction between the "Mixing Level" control, R_8, and the microphone level.



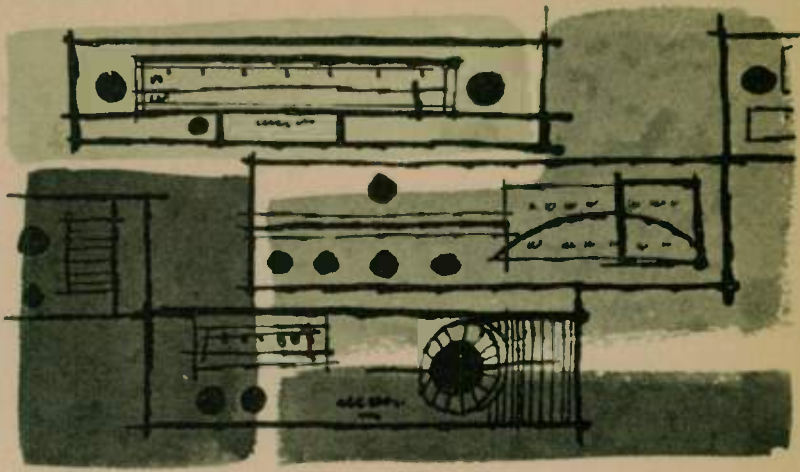
R_1, R_6, R_{11}—1 megohm, ½ w. res.
R_2—4700 ohm, ½ w. res.
R_3, R_4—220,000 ohm, ½ w. res.
R_5, R_{10}—100,000 ohm, ½ w. res.
R_7—10 megohm, ½ w. res.
R_8—500,000 ohm audio-taper pot (with switch, S_2)
R_9—470,000 ohm, ½ w. res. (see text)
C_1, C_4—.01 μf., 400 v. capacitor
C_2—25 μf., 25 v. elec. capacitor
C_3—135 $\mu\mu$f. ceramic capacitor
C_5—.047 μf., 400 v. capacitor
C_6—20 μf., 400 v. elec. capacitor
S_1—6-pole, d.t. switch (Centralab 2018 or equiv.)
S_2—S.p.s.t. switch (on R_8)
PL_1, PL_2—NE-2 neon bulb (NE-51 if bayonet lamp socket is used)
J_1—Phono jack
2—Stacked stereo record/playback head (Michigan Magnetics Type B or equiv. See text)
2—Matched erase heads (Michigan Magnetics Type SSB or equiv.)
V_1—ECC83/12AX7 tube

Fig. 2. Complete schematic diagram of the circuit modifications made.

Top rear view of the modified tape recorder showing the added controls.



new fm am tuner standard



By **DANIEL R. VON RECKLINGHAUSEN**
Chairman, IHFM Sub-Committee on Tuners
Chief Research Engineer, H. H. Scott, Inc.

Salient features of the Hi-Fi Institute's recently adopted standards for measuring AM/FM tuners.

THE Institute of High Fidelity Manufacturers has recently released its first standard, IHFM-T-100, entitled "Methods of Measurement for Tuners."

This standard represents a step forward in the techniques of tuner measurement. Previously, the only standards covering this field were those released by the Institute of Radio Engineers in 1947-48. These IRE standards were the best that could be devised at the time but later experience showed that certain necessary characteristics of tuner performance had not been included while other characteristics were measured in such a way that "good" test results did not necessarily mean "good" listening performance.

According to this new standard, a number of characteristics of either or both AM and FM tuners are to be measured: tuning range and frequency calibration, usable sensitivity, volume sensitivity, capture ratio, selectivity, amplitude modulation suppression, frequency response, distortion, spurious response, hum and noise, frequency drift, radiation, automatic frequency control, squelch control, loop antenna, and crosstalk.

Standard Conditions

All of these listed characteristics are to be measured under "standard" conditions. These conditions are:

Carrier frequencies: For FM tests, 90, 98, and 106 mc. and for AM tests 600, 1000, and 1400 kc. are standard test frequencies. Some tests are to be made using only one carrier frequency in which case it will be 98 mc. for FM and 1000 kilocycles for an AM tuner.

Since the settings of the various tuner controls will affect test results, certain standardized control positions have been specified. For example, the a.f.c. control, if present, is set for minimum control; the squelch control for maximum sensitivity; any tone controls for flattest response as indicated by panel markings; and volume control for maximum. Only in the case of tuners which incorporate extra stages of amplification, such as units equipped with phono-preamplifier or power-output stages, should the volume control be set for 20 db attenuation. This is to prevent clipping of the audio signal due to excessive amplification.

Input voltages, their measurement, and dummy antennas to be used for both AM and FM testing have been specified. These standards are diagrammed in Figs. 1 through 4.

The standard test modulation has also been specified. For most tests, the modulation frequency, except for some distortion and frequency response tests, is 400 cps. The standard modulation for FM is 100% or 75 kc. deviation while for AM this figure is 30%, except for distortion tests, where 90% is to be used.

Usable Sensitivity

If one wished to specify the most important characteristic of an FM tuner, a measurement of "usable sensitivity" would have to be made. The definition and measurement of usable sensitivity constitute the most important contribution made in the new standard. This term will be used generally within a few months to describe the sensitivity of tuners rather than the "quieting" figure as defined in the IRE standards and used up to now. The best definition and test method covering usable sensitivity of FM tuners is found in Section 6.03.02 of the new standard. Here is this section in its entirety:

"This test is performed at each of the standard test frequencies with the signal generator connected to the tuner under test through the standard 300-ohm dummy antenna. The signal generator should be frequency modulated with standard test modulation. The controls of the tuner shall be set to the normal control settings. The signal intensity should then be reduced to the least value which will produce a 30 decibel rise in indicated output with standard test modulation as compared with the indicated output with standard test modulation measured through a 400-cps null filter. This test serves to indicate the relative freedom of the tuner from objectionable internal receiver noise during pauses in modulation and receiver noise is least likely to be masked by modulation. This test also serves to indicate the relative freedom of the tuner from objectionable distortion during periods of maximum modulation.

"The results are expressed in microvolts."

What is new and different about this definition and how does it help the user to decide about the performance of a tuner?

First, this and all other measurements in the new standard are to be

If your recorder has provision for stereo playback then you will only have to add the extra erase head for the lower channel, the head switch, and the mixing level control. It will also be necessary to add sufficient capacity across your stereo preamp input to keep the bias oscillator from saturating it while recording. The coupling between the oscillator and preamp takes place in the recording heads. If you find this shunt capacitor noticeably affects the "highs," then change the value of your equalizing component to compensate for it. Don't let this discourage you, as it often involves changing a single component. Fig. 3 indicates the possible components making up your particular equalizing circuit. You can still have regular stereo operation by merely turning the "Mixing Level" control off and adding an amplifier for the second channel.

Procedure to be Followed

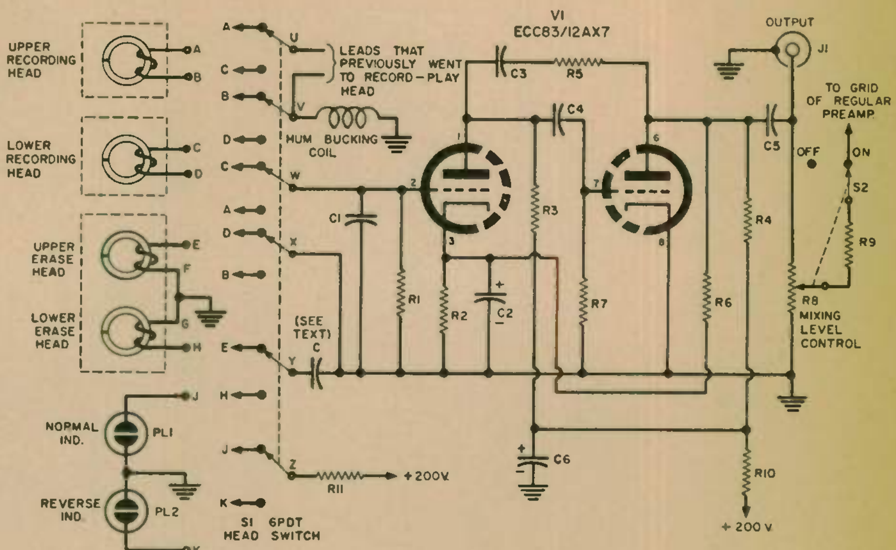
Although the following procedure is given for the *VM* Model 710 recorder, it will work equally well with most home recorders. These steps are given in the proper sequence for construction. *Important! Make certain the tools you use around the recording heads are not magnetized!*

First install the stereo record-playback heads and erase heads, lining them up as shown in Fig. 4. Mount the head indicator neons. The author merely drilled holes in the plastic head cover and cemented an NE-2 neon in each hole. These neons will probably last as long as the recorder and their use conveniently eliminates the need for lamp sockets, lens, etc. Don't let them protrude too far through the hole as they could get broken off.

Next mount the head switch, phono jack, and mixing level control in an accessible location, keeping them away from the electrical fields of the power transformer and motor. Now find a spot to mount the preamp, preferably near the original preamp and as far away from the power supply circuit as possible. Cut the required holes and put in grommets. Next, solder thin copper strips to the tube socket to serve as a shock-mount for the preamp, thus reducing the chances of microphonics. Mount the tube socket and then wire in the filaments, being careful to dress the leads away from pins 2 and 7.

Now label the head switch as shown in Fig. 2. This makes identification and wiring much simpler. Now wire all common points together, *i.e.*, "A" to "A," "B" to "B," etc. Next take the leads that went to your old record-play head and re-route them to points "U" and "V" of the head switch. Wire in the newly installed record-play heads according to details shown in Fig. 2. Remember to ground only one end of the shield. One of the leads that went to the erase is grounded. Take the other "hot" lead and re-route it to point "Y" of the switch. Wire up points "E" and "H" to the same "E" and "H" points on the head switch. Likewise wire in the neon indicators to points "J" and "K" of the switch. Wire in the voltage to light the neons to point "Z," adding the 1-megohm resistor in series.

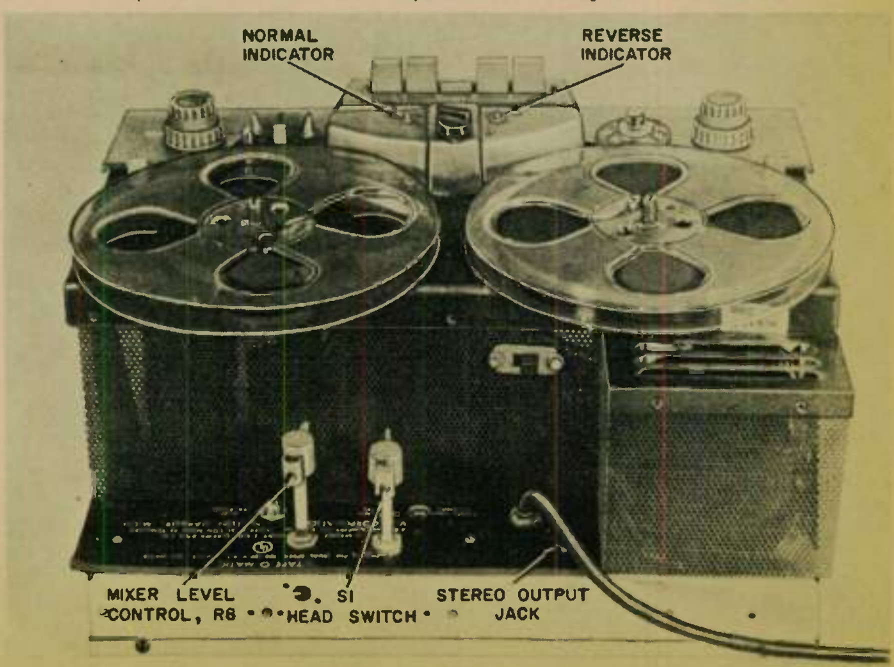
Wire in the other components, keeping all leads as short and direct as possible and using shielded cable for longer leads. Regular plastic-covered shielded microphone cable was used in the author's unit with very good results. The output of your new preamp, coming off the .047 µf. (C_5) capacitor, is to be wired to either the first or second amplifier stage of your regular recorder circuit. The author used the second stage, pin 7 of the normal preamp, as it provided sufficient gain. If more gain is required, you can use the first stage and change R_9 from 470,000 ohms to 1 megohm to minimize interaction between the "Mixing Level" control, R_8, and the microphone level.



R_1, R_6, R_{11}—1 megohm, ½ w. res.
R_2—4700 ohm, ½ w. res.
R_3, R_5—220,000 ohm, ½ w. res.
R_4, R_{10}—100,000 ohm, ½ w. res.
R_7—10 megohm, ½ w. res.
R_8—500,000 ohm audio-taper pot (with switch. S_2)
R_9—470,000 ohm, ½ w. res. (see text)
C_1, C_4—.01 µf., 400 v. capacitor
C_2—25 µf., 25 v. elec. capacitor
C_3—135 µµf. ceramic capacitor
C_5—.047 µf., 400 v. capacitor
C_6—20 µf., 400 v. elec. capacitor
S_1—6-pole, d.t. switch (Centralab 2018 or equiv.)
S_2—S.p.s.t. switch (on R_8)
PL_1, PL_2—NE-2 neon bulb (NE-51 if bayonet lamp socket is used)
J_1—Phono jack
2—Stacked stereo record/playback head (Michigan Magnetics Type B or equiv. See text)
2—Matched erase heads (Michigan Magnetics Type SSB or equiv.)
V_1—ECC83/12AX7 tube

Fig. 2. Complete schematic diagram of the circuit modifications made.

Top rear view of the modified tape recorder showing the added controls.



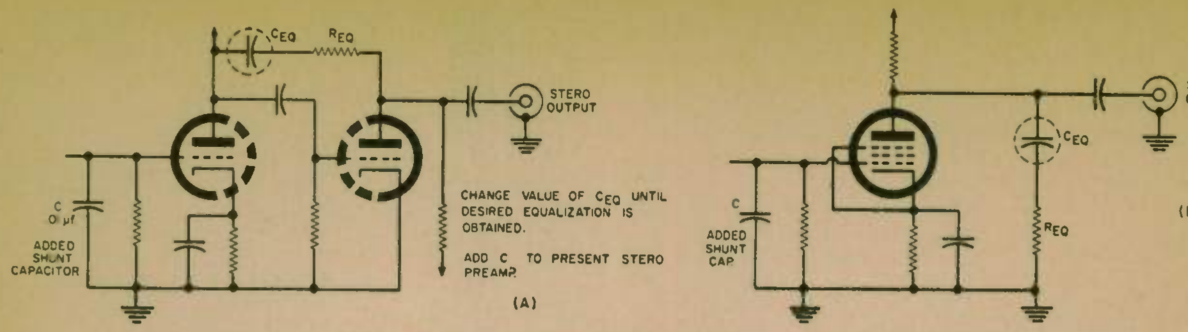
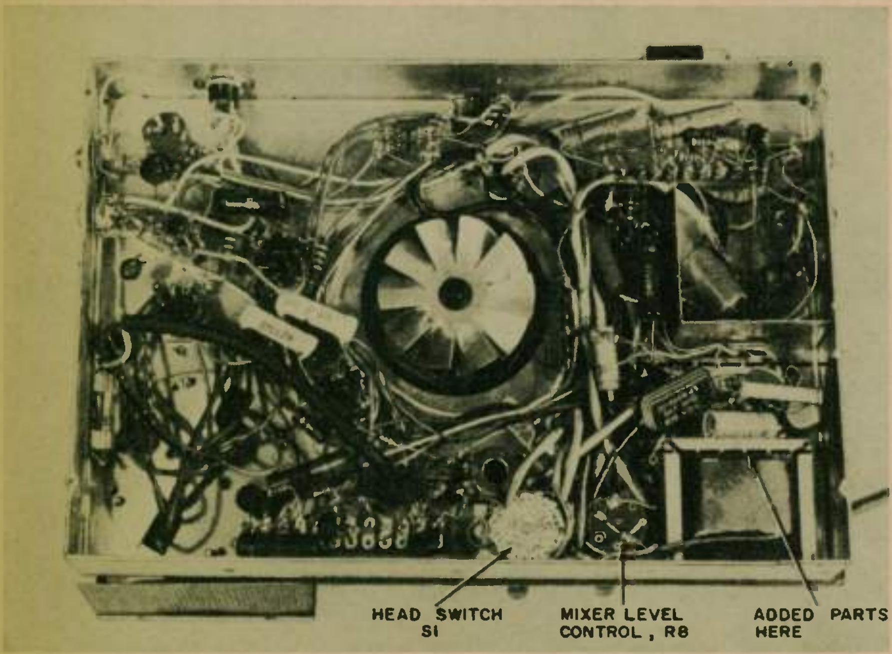


Fig. 3. Circuits showing equalization components of two basic tape amplifiers. Refer to text for details.



Under-chassis view showing the area where the added components have been located.

In some recorders the "External Speaker" jack is inoperative in the "Record" position. If this is the case, it will be necessary to rewire the output jack direct from the output transformer through a simple s.p.s.t. switch to turn it on and off. Fig. 5 shows how this can be accomplished. The reason both terminals of the stereo heads are switched in the author's unit is the presence of a hum-bucking coil in the normal recording track. If your recorder has one side of the record head grounded, then a four-pole, double-throw switch can be used in place of the six-pole unit.

As previously stated, only one end of the shield should be grounded. If you were shielding only to prevent interaction from electrostatic fields, then the number of times and the places it was grounded would not matter. However, a slight ground differential may be present between the deck and amplifier chassis. If both ends of the shielded pair are grounded, then a closed loop exists and the shield now becomes a conductor for these currents, coupling hum and noise into the sensitive preamps by means of electromagnetic induction. This situation can develop even though ohmmeter readings do not show any appreciable resistance between the two points.

Head Adjustment

Now that the wiring is completed, we are ready to line up those newly installed heads. There are precise procedures for doing this and those who have alignment tapes, meters, etc. are urged to use them as directed. The recording heads selected by the author require 1 ma. of bias current with 25 µa. of audio for optimum performance. The following method is suggested for those who do not have access to regular alignment equipment and it will give satisfactory results.

Turn the "Mixing Level" control, R_8, fully off. Put the "Head Switch" (S_1) in "Normal" position. See Fig. 1. With a recorded tape containing good highs and lows, simply orient the head for maximum playback quality, paying particular attention to the extreme highs. Now adjust the erase heads so they are physically in line with the record-playback heads, as shown in Fig. 4. As the Type "SSB" erase heads require only 17 ma. of erase current for 50 db erasure, it will be necessary to check this current in order to avoid damaging the heads.

Since measuring high-frequency currents can lead to complications without special meters, here is a simple, yet practical, method of reaching the proper erase current while avoiding damage to the heads. Start out by shunting a relatively large capacitor (.01 µf.) across the output of the erase oscillator as it enters the head switch. This capacitor is designated as C in Fig. 2. Turn the "Mixing Level" control off and S_1 to "Normal" position. Put on a recorded tape that you won't mind erasing and set the recorder to the "Record" position with the volume control at minimum setting. Now play the tape back and see how clean it was erased. Gradually decrease the value of this shunt capacitor, each time testing the erasing qualities on the recorded tape until erasure is complete even when a relatively high level of recording has been made.

If the lower track is partially erased, recheck the relative position of the erase heads to the recording heads. If your erase heads are mounted separately, it will be necessary to check inter-channel erasing with S_1 in the "Reverse" position, each time re-positioning the lower erase head for the best erasure without affecting the upper track.

Trying It Out

Now you are ready to try it out. If

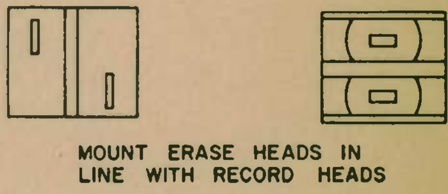
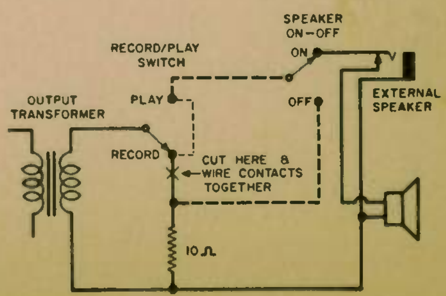


Fig. 4. Lineup of the tape heads used.

Fig. 5. Wiring of the output circuit.



an odd number of parts are to be recorded, put S_1 in the "Normal" position and for an even number of parts start off in the "Reverse" position. By doing this you will always end up with the completed recording on the normal track, thus the tape can be played on any recorder. Always record the first part with the "Mixing Level" control (R_8) turned off. After recording is completed for the first part using mike, radio, phono, etc., rewind the tape to the starting position. Turn S_1 to the opposite track and turn R_8 full on. With headphones in the "External Speaker" jack, and the recorder volume control about half way on, start recording, but this time harmonizing with the first part. You can hear the two parts blended together in the headphones and with a little experience you will soon know the proper levels needed to turn out good quality recordings.

If the first part is too loud, change the setting of R_8. Your regular volume control changes the level of both tracks. Rewind the tape to the starting point again, change the setting of S_1, leaving R_8 in the same position. Record the third part, harmonizing with the other parts. Continue this procedure, each time rewinding the tape to the starting position and changing the setting of S_1. When you are all through, turn R_8 full off and play back the tape in the regular manner. A form of "pseudo stereo" can be obtained by putting an amplifier and speaker on the output of the second channel.

You can also play regular stereophonic tapes by simply adding an amplifier to the output of the second channel and turning the "Mixing Level" control full off. If R_8 is turned on, you can play both tracks through your regular tape amplifier and enjoy the good quality of stereophonic tapes—even monophonically.

You will find this addition to your recorder a very worthwhile effort and its applications are limited only by your own ingenuity and imagination. This scheme works very well with organ-piano duets as well as vocals. You can even sing your "barber shop" quartet numbers.

There are a number of extras that some might like to add, such as level indicators for each channel, etc., but with what's given here you can have lots of fun and—who knows—you may discover that you have "hidden talents."

To the author's delight, he has found that the more parts, the less significant are the mistakes of the single parts—in other words—a trio seems to sound better than a solo.

Have fun, but remember one's talents aren't always adequately appreciated by others and your neighbors may look upon your new "pride and joy" as being nothing more than a monstrously amplified device for magnetically multiplexing musical noise and aggravating others! Don't let this discourage you as they just don't know the fun they are missing.

Bulk Eraser for Magnetic Tape

By WILLIAM QUITMAN WOLFSON, M.D.

WITH most inexpensive recorders, even when the instrument is new, it is difficult to erase completely earlier recordings from previously recorded tape, yet it is essential to do so for optimum results in certain problems. Even very slight background will markedly impair a high-fidelity recording. If the recording apparatus has been used much, the erase efficiency is generally not maximal because of worn pressure pads or inexact alignment of the head horizontally or with respect to azimuth, defects not so obvious as if they had occurred with the record or playback head. However, even if these factors are carefully controlled, poor erase may ruin a critically important tape if the line voltage happens to be low when the recording is made or if the tape happens to be from a lot which is intrinsically difficult to erase. During 1954, the writer purchased some 1-mil Mylar tape which was not notably reduced in intensity by one passage across the erase heads of any of a half-dozen different American and European home-type recorders.

The solution to these problems is the use of a so-called "bulk eraser" before the recording is made. A properly designed bulk eraser will, in a matter of seconds, completely remove previous recordings from a reel of tape and leave it with a background level less than that which it had when new. Unfortunately, many commercially available erasers are expensive enough to discourage their wide use by amateur home recordists. Fortunately, however, home construction of an adequate bulk eraser is almost ridiculously simple and inexpensive, providing the right starting material is used.

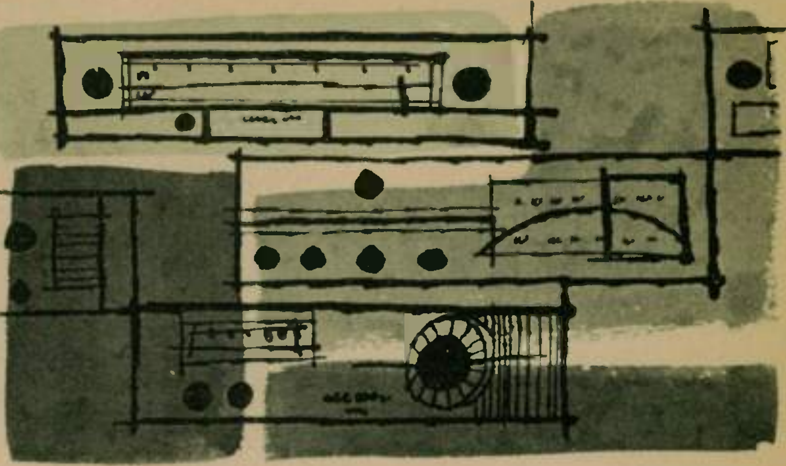
Purchase a receiver or transmitter power supply filter choke of the commonly used 5 to 25 henrys inductance, rated at 75 to 200 ma. capacity. If bought surplus, such chokes may cost as little as 70 cents and, in many cases, should not cost more than several dollars. Remove the case of the choke. Examine the laminations. These will be found to consist of two groups: (a) a series of E-shaped laminations with a coil wound on the center bar of the E and (b) a series of straight laminations which close the open parts of the E. The latter are generally held against the E only by varnish. Using a screwdriver to direct the force properly, knock off the straight laminations and discard them. There now remains the coil wound on the center bar of the E-shaped laminations. Fasten these E-shaped laminations together tightly with two nuts and bolts; most chokes already have holes which permit this to be done without drilling. Tape the coil down securely with electrical insulating tape and attach its two leads to an ordinary line cord and 117-volt male plug. A convenient additional feature is to mount a press-to-make switch on the unit. It is connected in one side of the 117-volt line and permits the unit to be plugged in continuously with current flowing only when the switch is pushed.

Erasing Technique

The effective erasing field is located at the open ends of the E and, like any electromagnetic field, it falls off in intensity in proportion to the square of the distance from this region. This means that, while being used, the unit should be closely applied to the reel of tape being erased. In practice, the eraser is plugged into the 117-volt socket and passed slowly over the reel so that every area on the reel passes directly under one of the two open gaps in the E. A simple extra precaution against failure to erase some small area is to pass the eraser successively over both faces of the tape reel, although this is unnecessary if the first face is covered carefully. After this process is completed, slowly withdraw the eraser to about two feet from the reel before turning it off; this is a precaution against recording the self-induction current surge which occurs after the current is turned off. The coil is carrying considerably more current than its rated continuous capacity and therefore the unit should be operated only intermittently. With the prototypes, up to ten reels were erased without excessive heating. Ample warning against excessively prolonged use is given by the obvious rise in coil temperature if the unit is used for a long period.

The only parts needed are the choke, two nuts and bolts, one 117-volt male plug and cord and some insulating tape; all at a cost of only a few dollars.

new fm am tuner standard



By **DANIEL R. VON RECKLINGHAUSEN**
Chairman, IHFM Sub-Committee on Tuners
Chief Research Engineer, H. H. Scott, Inc.

Salient features of the Hi-Fi Institute's recently adopted standards for measuring AM/FM tuners.

THE Institute of High Fidelity Manufacturers has recently released its first standard, IHFM-T-100, entitled "Methods of Measurement for Tuners."

This standard represents a step forward in the techniques of tuner measurement. Previously, the only standards covering this field were those released by the Institute of Radio Engineers in 1947-48. These IRE standards were the best that could be devised at the time but later experience showed that certain necessary characteristics of tuner performance had not been included while other characteristics were measured in such a way that "good" test results did not necessarily mean "good" listening performance.

According to this new standard, a number of characteristics of either or both AM and FM tuners are to be measured: tuning range and frequency calibration, usable sensitivity, volume sensitivity, capture ratio, selectivity, amplitude modulation suppression, frequency response, distortion, spurious response, hum and noise, frequency drift, radiation, automatic frequency control, squelch control, loop antenna, and crosstalk.

Standard Conditions

All of these listed characteristics are to be measured under "standard" conditions. These conditions are:

Carrier frequencies: For FM tests, 90, 98, and 106 mc. and for AM tests 600, 1000, and 1400 kc. are standard test frequencies. Some tests are to be made using only one carrier frequency in which case it will be 98 mc. for FM and 1000 kilocycles for an AM tuner.

Since the settings of the various tuner controls will affect test results, certain standardized control positions have been specified. For example, the a.f.c. control, if present, is set for minimum control; the squelch control for maximum sensitivity; any tone controls for flattest response as indicated by panel markings; and volume control for maximum. Only in the case of tuners which incorporate extra stages of amplification, such as units equipped with phono-preamplifier or power-output stages, should the volume control be set for 20 db attenuation. This is to prevent clipping of the audio signal due to excessive amplification.

Input voltages, their measurement, and dummy antennas to be used for both AM and FM testing have been specified. These standards are diagrammed in Figs. 1 through 4.

The standard test modulation has also been specified. For most tests, the modulation frequency, except for some distortion and frequency response tests, is 400 cps. The standard modulation for FM is 100% or 75 kc. deviation while for AM this figure is 30%, except for distortion tests, where 90% is to be used.

Usable Sensitivity

If one wished to specify the most important characteristic of an FM tuner, a measurement of "usable sensitivity" would have to be made. The definition and measurement of usable sensitivity constitute the most important contribution made in the new standard. This term will be used generally within a few months to describe the sensitivity of tuners rather than the "quieting" figure as defined in the IRE standards and used up to now. The best definition and test method covering usable sensitivity of FM tuners is found in Section 6.03.02 of the new standard. Here is this section in its entirety:

"This test is performed at each of the standard test frequencies with the signal generator connected to the tuner under test through the standard 300-ohm dummy antenna. The signal generator should be frequency modulated with standard test modulation. The controls of the tuner shall be set to the normal control settings. The signal intensity should then be reduced to the least value which will produce a 30 decibel rise in indicated output with standard test modulation as compared with the indicated output with standard test modulation measured through a 400-cps null filter. This test serves to indicate the relative freedom of the tuner from objectionable internal receiver noise during pauses in modulation and receiver noise is least likely to be masked by modulation. This test also serves to indicate the relative freedom of the tuner from objectionable distortion during periods of maximum modulation.

"The results are expressed in microvolts."

What is new and different about this definition and how does it help the user to decide about the performance of a tuner?

First, this and all other measurements in the new standard are to be

made with a 300-ohm antenna since most antennas have this impedance. This insures that the same yardstick is used on all tuners. It may not be generally realized but measurements performed at 75 ohms will give results, in microvolts, which are half those obtained with 300-ohms impedance although performance of the tuner is not one hair better. The reason is that the same power is required by the tuner for a given performance and this power is simply the square of the voltage at the antenna terminals divided by the antenna impedance.

Second, a 300-ohm dummy antenna will have to be used and the input voltage to the tuner measured by replacing the tuner with a voltmeter having an input impedance of 300 ohms. This input voltage is the same as the "soft microvolts" used before except that here the 300-ohm impedance is specified. This will insure that tuners are designed for best possible performance with a standard 300-ohm antenna rather than for the lowest microvolt figure, irrespective of impedance.

Third, if a tuner should have more than 3% distortion it would be impossible to obtain a usable sensitivity figure no matter how high the input to the tuner might be. In this way, the customer is assured of getting a tuner with no more than 3% distortion at any input voltage equal to, or higher than, the usable sensitivity test input. Generally, this distortion figure will run considerably lower. Distortion is measured at 75 kc. deviation at a 400-cps rate which is standard test modulation in the IHFM Standard as well as 100% modulation for an FM broadcast station. In this type of measurement care must be taken to insure that the null filter has a high-impedance input to avoid attenuating the frequencies far removed from 400 cps. A low-impedance filter may cause distortion in the audio circuits of the tuner and a filter which attenuates the second and higher harmonics of 400 cps will give a distortion figure which is much lower than the actual distortion.

Volume Sensitivity

Closely linked to the measurement of usable sensitivity is the measurement of "volume sensitivity," as covered in Section 6.03.03. This covers the r.f. input required to produce an audio output 20 db lower than that produced by a 100,000 μv. signal. High-quality, high-gain tuners will measure zero microvolts in this test while tuners with insufficient amplification or limiting may show a microvolt figure which is greater than that obtained in the usable sensitivity test. For this purpose, Section 6.03.03 states:

"The rated sensitivity of a tuner shall be equal to the highest number of microvolts obtained in all tests of Sections 6.03.02 and 6.03.03. . . ." and these tests to be made at 90, 98, and 108 mc. All this should insure that tuners with rated sensitivities in low microvolt figures should have good sensitivity, low distortion, and adequate audio output irrespective of signal strength.

As users and designers of FM tuners know, sensitivity is not the only important tuner characteristic although it is often given as the major specification. Equally important is the tuner's ability to reject interference.

Capture Ratio Test

The capture ratio test is a measure of the tuner's ability to reject unwanted signals occurring at the same frequency to which the unit is tuned. This test shows the inherent effectiveness of the detector, limiter, and automatic volume control circuits. For this test, two signal generators are required, one of which is modulated 100% and the other unmodulated. A dummy antenna, which permits two signal generators to be connected to a single tuner, is used in this test.

To measure the rated capture ratio of the tuner the modulated generator is set to produce a 1000 μv. input to the tuner while the output of the unmodulated generator is advanced until the audio output of the tuner falls 1 decibel. The output of the generator is then recorded. Next the output control of the unmodulated signal generator is advanced until the audio output of the tuner falls a total of 30 decibels. This figure is again recorded.

The ratio of the two values thus recorded is converted to decibels and this figure is divided by 2. This result is defined as the "db capture ratio" for 100% modulation. This is the ratio of desired or undesired signal required for 30 db suppression of the undesired signal. This test is extremely important since there are a number of stations in the northeastern section of the United States which are assigned the same frequency but are physically separated by less than 100 miles. If a tuner has a low capture ratio, then one or the other of these stations can be received in some areas without noticeable interference by simply rotating the antenna to pick up the signal from the desired station.

Selectivity Test

Equally important is the tuner's ability to reject interference and signals which differ in frequency from that of the desired signal. Part of this measurement is performed in the selectivity test.

Test conditions are the same as those described in the capture ratio test except that the interfering signal generator is separated in frequency from the desired signal by two standard channels (400 kc.). The signal generator delivering the desired signal is set to produce an input of 100 μv. to the tuner and the audio output voltage with 100% modulation is recorded. Then the modulation of the signal generator is switched off. The interfering signal produced by the other signal generator is 400 kc. removed from that of the desired signal. This interfering signal is modulated with 400 cps, 75 kc. deviation and the output of this signal generator is advanced until the audio output of the tuner has risen to a value 30 db below the previously determined audio output. The two signal generator output voltages are recorded and converted to a ratio which is then converted to decibels. This decibel figure

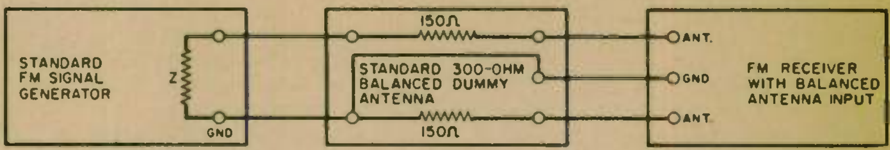


Fig. 1. Standard 300-ohm balanced dummy antenna and the method of connection.

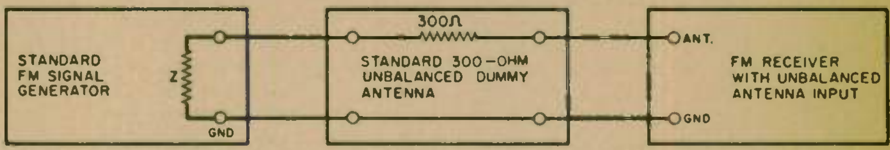


Fig. 2. Standard 300-ohm unbalanced dummy antenna and the method of connection.

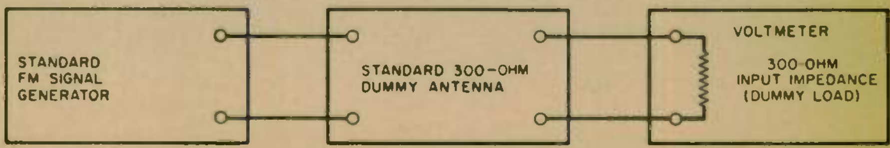
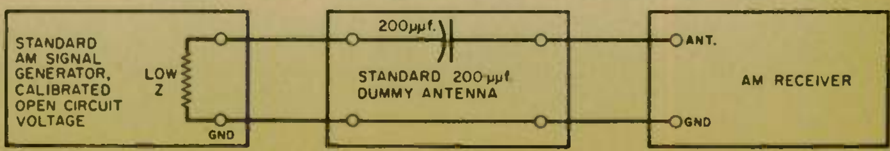


Fig. 3. Method of measurement of input signal intensities and connection method.

Fig. 4. The standard 200-μμf. dummy antenna is between generator and receiver.



gives the selectivity of the tuner.

The same procedure is followed in the case of AM tuners except that 30% amplitude modulation is used.

It should be noted that all of these tests of selectivity and capture ratio have to be performed at the three standard carrier frequencies and the poorest figure obtained becomes the "rated performance" of the tuner.

Spurious Response

The third type of measurement which indicates the capability of a tuner in rejecting interference is the spurious response test. Here the tuner is set to each of the standard carrier frequencies and a signal generator with sufficient output voltage tuned over a wide frequency range. At each frequency where the tuner shows an audio output the signal generator output voltage is adjusted so that the measurement procedure of the usable sensitivity test is followed. This voltage with the usable sensitivity test input voltage is converted to a ratio and then expressed in decibels. The number of decibels gives the amount of spurious response rejection by the tuner. A number of these spurious response frequencies may be found, possibly due to forms of r.f. distortion within the tuner. Two particular ones will have to be recorded separately: image response and i.f. response. These frequencies may turn out to be 10.7 mc. for i.f. response and equal to the tuned frequency plus 21.4 mc. for the image response frequency of the tuner.

Other Tests

The other tests normally performed on tuners are self-explanatory and it is merely a case of following the conditions specified in the standard. Some of these tests were made in the past so it is only necessary to adopt the proper reference levels in order to meet the new IHFM standard. The tests which have been described are basically new tests and their general acceptance will be a matter of gradual education.

A copy of the new standard, giving complete details, may be obtained from the Institute of High Fidelity Manufacturers, Inc., 125 East 23rd Street, New York 10, N. Y. The price of IHFM-T-100 is $1.00.

The author wishes to thank the members of the IHFM Standards Committee, Sub-Committee on Tuners: Lawrence B. Arguimbau, *McIntosh Laboratories;* Joseph N. Benjamin, *Bogen-Presto Division* of the *Siegler Corporation* (president of IHFM); Leonard Feldman, *Madison Fielding Corporation;* Robert F. Furst, *Harman-Kardon, Inc.;* A. A. Hart, *Precision Electronics;* Allen Holstrom, *Stromberg-Carlson;* Joseph Levitsky, *Bogen-Presto Division* of the *Siegler Corporation;* F. L. Mergner, *Fisher Radio Corporation;* Edward Miller, *Sherwood Electronic Laboratories;* and Richard Shottenfeld, *Pilot Radio Corporation* for their cooperation and assistance in establishing these new standards.

Experiments in Stereo for P.A. Systems

By **JACK THORNTON**

RECENT comparison tests with stereophonic music systems suggested a re-investigation of a two-channel technique in public-address system work. The results were quite interesting and the idea seems worthy of further experiment by those working with auditorium amplifier setups. Such a system uses two separate sound channels set up in a "left" and "right" relationship which is essentially the same as that used in stereo music recording and playback.

Auditorium public-address systems usually use a loudspeaker enclosure at either side of the stage. The two sound sources obviously give better sound distribution and coverage. But the system remains monophonic because both speakers are fed by a common microphone and amplifier.

Consider a person speaking from a stage *without* a sound system. The audience experiences the normal acoustical effect, that is, when the speaker turns slightly to his left, the audience in that direction hears the voice projected a little more toward them. The audience the other way hears the voice diminish slightly. Even though this effect may not be too pronounced, natural sound does depend upon it.

When the same person uses the ordinary single-channel sound system, the loudspeakers override his actual voice so that listeners hear only what comes over the p.a. Now when he turns slightly, his voice fades in *both* directions as he goes "off mike" and the sound diminishes in both speakers. The sense of movement and liveness is impaired. Of course there is the visual sense of direction because the audience can see the person and his movements. But they still hear the sound through a monophonic system. A dual-channel sound system seems to offer one method of improvement. This holds true even though the source of sound moves very little.

First tests of p.a. stereo were run in adjoining radio studios with a window between them. A curtain could be drawn for comparison between impressions when the observers could or could not see the speaker. Matched microphones fed companion amplifiers in studio A. Matched speakers were placed eight feet apart in studio B. A switching arrangement allowed single- or dual-channel operation, with both speakers operating either way. Observers had no difficulty identifying the "stereo" setup, even with the curtains drawn. The person talking was twelve inches from the mikes. Mike separation varied from two inches to two feet. The larger separations gave exaggerated effects. About ten inches seemed the most natural in that particular studio.

More checks were run in an auditorium. Microphones were *Shure* 556 cardioids to allow somewhat greater mike-to-subject distance without audio feedback. The systems were balanced by having an observer at the center of the auditorium indicate when the sound seemed to come from the middle of the stage.

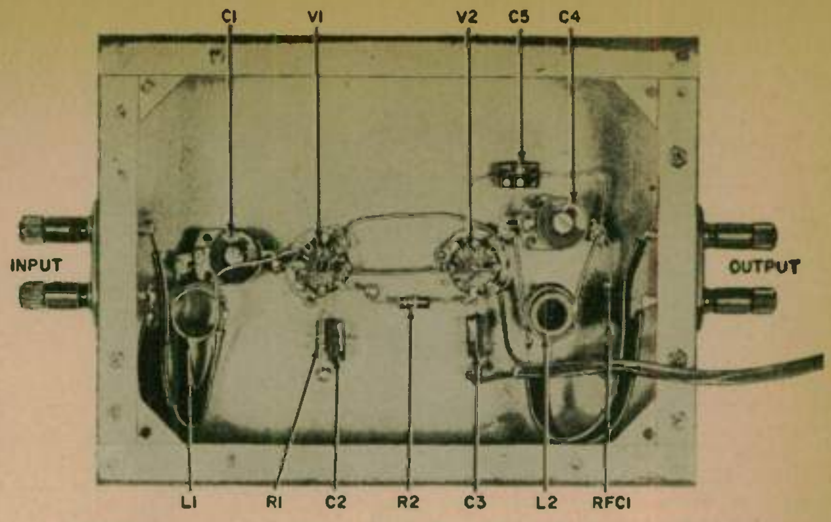
With speech as the program material, trial and error finally placed the microphones on the speaker's rostrum twelve inches from the voice, separated from each other ten inches, and angled in slightly toward the sound source. The closer the mikes are to the voice, the closer they must be to each other if a "ping-pong" situation is to be avoided. (It's quite an effect to have the speaker turn his head slightly and seem to jump thirty feet to the other side of the stage!)

If the microphones are too close to the voice, most of the stereo effect is lost since any movement will put the speaker off both mikes and you'll be back to a monophonic system. Unless the speaker has a weak voice or the room is a bad one for feedback, the greater mike distance will give better sound anyway.

When music is the program material, both the volume of sound and the size of the group producing it must be considered. A vocal or instrumental solo usually allows greater mike distance: two or three feet. Mike separation is accordingly greater. As the musical group becomes larger so must microphone distances become greater. Large orchestras or choirs seldom need amplification and if a sound system is used with these, it's usually for a soloist and the solo settings can be used.

Operators who wish to duplicate and expand these experiments could use this guide: microphones, amplifiers, and loudspeakers should be as identical as possible. Identical models may not be matched; one unit may be new, the other a veteran of rough service. Tone controls are set the same for both units, although some compensation for unmatched units can be obtained by tone setting variations. With mikes about twelve inches from a voice and separated about ten inches, balance the two channels by having an observer indicate when the sound seems to come from the speaker. Then vary the mike distance and separation for best effect with the program material to be used.

Such a setup is an interesting sidelight on the stereo boom and, although it won't hide poor p.a. systems or practice, it offers a worthwhile improvement in the naturalness of sound reproduction.

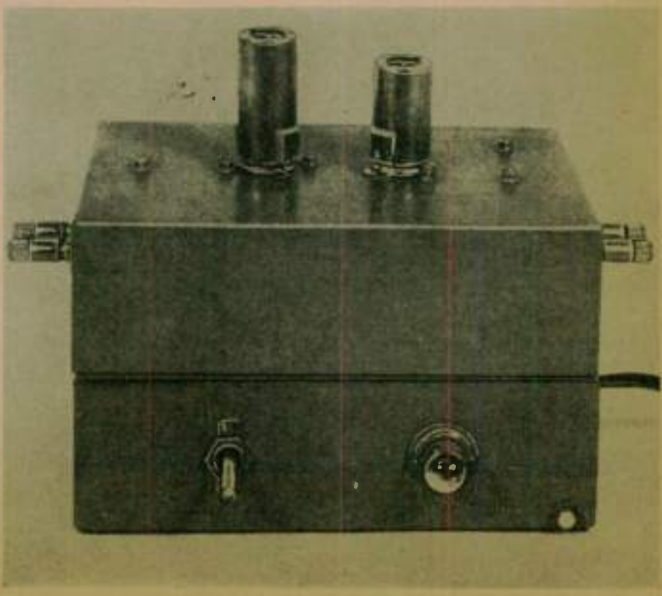


Under-chassis view of r.f. section. Short, direct wiring should be employed here.

Simple Booster For FM Tuners

By FRANCIS A. GICCA
Senior Engineer, Raytheon Co.

Design and construction of cascode front end that increases sensitivity by 20 db. The unit can be built for only about $15.


The r.f. chassis of the booster is atop the power-supply chassis.

PERHAPS nothing can be more annoying than insufficient FM receiver sensitivity, particularly when you are planning to tape-record a live concert from a distant station only to find that you can't receive the station very well at all.

Why don't more FM receiver manufacturers build in sufficient tuner sensitivity so that this problem will never occur? They can, but at a price. Building high gain into the front end of an FM tuner is a difficult engineering job. First, the r.f. amplifier must be wideband since it has to amplify all frequencies in the FM band from 88 to 108 megacycles. Achieving high gain over such a wide bandwidth is almost impossible. It must be remembered that a vacuum-tube amplifier is limited in gain, that is, even with optimum circuit design, there is a *maximum* amount of gain that can be extracted from a vacuum tube. The better the tube, the higher will be its maximum available gain.

Unfortunately, a high-gain tube also generally has higher internal capacitance between its electrodes. This interelectrode capacitance means that as the input frequency is increased the gain of an amplifier begins dropping due to the shunting of these capacitors at high frequencies. As a result, vacuum tubes have a maximum achievable bandwidth because of internal stray capacitance. A good vacuum tube, therefore, is characterized by a high maximum gain plus high bandwidth capabilities. In order to compare the relative performance of tubes, a good "figure of merit" is the product of the maximum obtainable gain and maximum bandwidth. This figure of merit is generally known as the "gain-bandwidth product" (F_t). The better the tube, the higher its gain-bandwidth product. Table 1 lists the gain-bandwidth products for several popular pentodes.

Note that the 6AC7 has a gain-bandwidth product of 562 million. If the 6AC7 were to be used as an r.f. amplifier in an FM tuner, what is the maximum gain that can be expected? The FM band covers 88 to 108 megacycles; a bandwidth of 20 megacycles. The maximum gain that can be expected is 562 million divided by 20 million, or 28.1. This is a maximum gain of 29 db. In production units, such gain could never be achieved consistently, therefore a gain of about 20 db would be normal with a 6AC7.

Gain is not the only requirement of an r.f. amplifier, it must also be relatively noise free. Here again, noise requirements are in conflict with high gain. Pentodes are noisier than triodes, but pentodes have the higher gain. Since low noise is vital in an FM tuner, a compromise is generally necessary in the choice of input tubes and lower-noise types are favored over higher gain, but noisier, types. The result is that r.f. amplifier gain is generally limited to between 10 and 15 db.

In order for a tuner manufacturer to build high sensitivity into his product it would be necessary to cascade several r.f. amplifying stages. This is expensive and may make his tuner uncompetitively priced. High sensitivity is needed only occasionally, therefore, the manufacturer compromises and designs a tuner with reasonable sensitivity at a reasonable price. On distant stations, then, many FM tuners will lack sensitivity. Even weak local FM stations may be received marginally as evidenced by background noise or annoying "snaps" and "pops" due to automobile ignition systems.

What can you do to increase the sensitivity of your FM tuner? There are two possible solutions. You can install a high-gain FM antenna or you can add an external r.f. amplifier to boost the strength of the signal before it enters the tuner.

FM antennas can improve receiver sensitivity, but by only a limited amount. Why do antennas have gain? They are, after all, completely passive devices and therefore cannot add any additional power to the incoming signal. Antennas have "gain" because they are directional and capable of favoring signals from a given direction to the exclusion of signals from all other directions. A simple folded dipole can achieve a gain of between 3 and 4 db with moderate directionality. A six-element yagi can provide gains between 6 and 9 db at the expense of considerable directionality. These gains, although relatively low, may be adequate for many marginal stations. However, because of the directionality of antennas, all marginal stations must be located in the same general direction from the receiver in order to benefit from the antenna's gain. If this is not so, then the antenna will have to be rotated to face each station. Perhaps an FM booster is a more practical and economical solution.

FM Booster Design

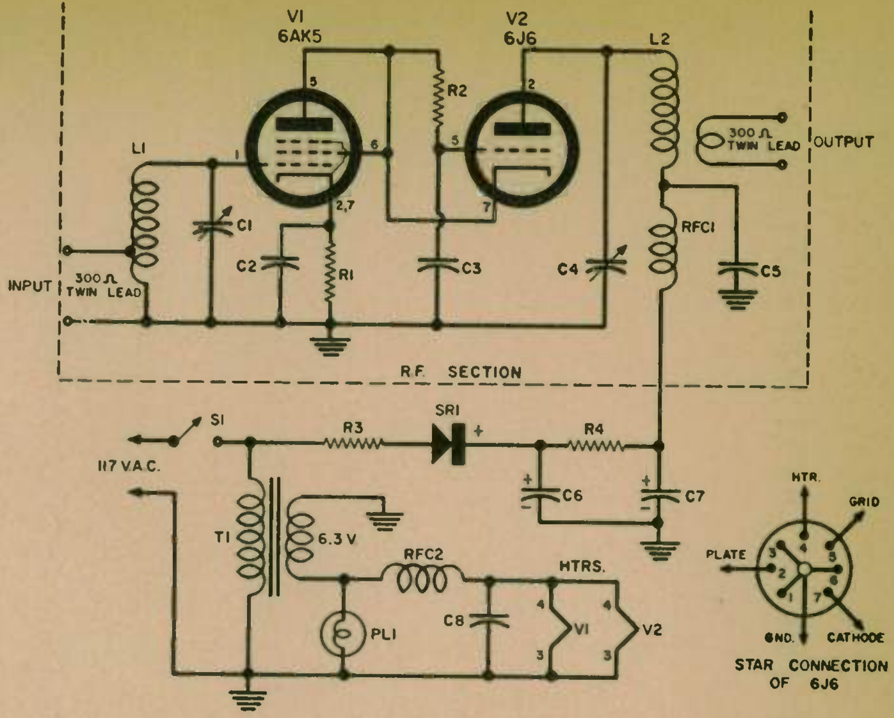
Keeping in mind what has been said about r.f. amplifiers, an FM booster can be designed which will yield higher sensitivity than any antenna system. Since low noise is of paramount importance, the circuit chosen must have inherently good noise characteristics. The triode exhibits far superior noise performance than a pentode but, as mentioned previously, it has lower gain. In the design of television front ends this same problem was met and solved with the development of the "cascode" amplifier. Basically, the cascode amplifier has the gain of a pentode with the low-noise characteristics of a triode. The cascode amplifier consists of two triodes, the first connected grounded-cathode and the second connected grounded-grid. Since the cascode amplifier offers the desired characteristics of high gain and low noise, it was decided to choose this configuration for the FM booster.

If a 6AK5 (connected as a triode rather than a pentode) and a 6J6 are connected in cascode, a gain-bandwidth product of about 500 million can be expected. Note that this compares favorably with the best pentode in Table 1, the 6AC7 with its gain-bandwidth product of 562 million.

TUBE	F_t
6AB7	385
6AC7	562
6AG7	385
6L6	236
6SJ7	127
6SK7	154
6V6	195

Table 1. Gain-bandwidth products ($\times 10^6$) are shown for several popular pentode tubes.

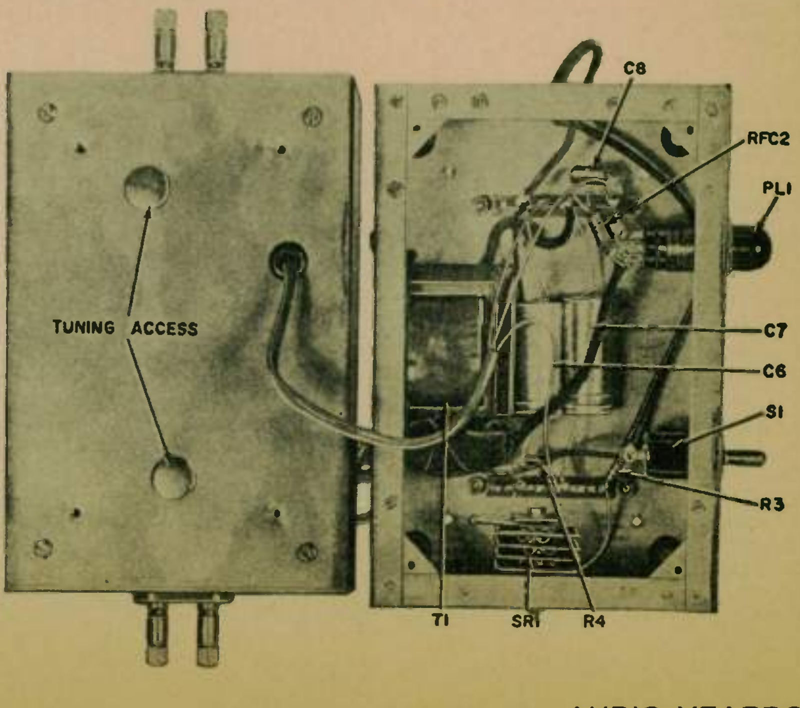
What gain would be desirable for our FM booster? If a gain of at least 20 db could be achieved, this would make the booster far superior to any antenna system and certainly worth considering. We must remember that the gain-bandwidth product specifies the *maximum* performance that can be expected, therefore, a safety factor should be included to account for the losses that will occur in an actual circuit. Let's arbitrarily try for a gain of 40 db, knowing that this will allow a more than adequate safety factor. Forty decibels is a gain of 100. Since the gain-bandwidth product is about 500 million, the maximum bandwidth that can be obtained is 5 megacycles (F_t divided by 100). This means that



R_1—68 ohm, ½ w. res.
R_2—51,000 ohm, ½ w. res.
R_3—15 ohm, ½ w. res.
R_4—1500 ohm, ½ w. res.
C_1, C_4—6 to 12 μμf. trimmer or variable (E. F. Johnson Type 160-107, see text)
C_2, C_3, C_5, C_8—510 μμf., 500 v. mica capacitor
C_6, C_7—20 μf., 150 v. elec. capacitor
L_1—.3 μhy. tapped coil. 4½ t. #16 en. wire wound on ½-inch coil form. Turns spaced to 1-inch over-all. Tap at 1½ t. from bottom end.
L_2—.3 μhy. output tank. 4½ t. #16 en. wire wound on ½-inch coil form. Turns spaced to 1-inch over-all. 1 t. output link wound between bottom wire turns
RFC_1—10 mhy., 75 ma. r.f. choke (National R-300 or equiv.)
RFC_2—.36 μhy., 2 amp r.f. choke (National B20407-4 or equiv.)
SR_1—20 ma., 130 v. selenium rectifier (Federal 1159 or equiv.)
S_1—S.p.s.t. toggle switch
PL_1—6 v. pilot light
T_1—Fil. trans., 6.3 v. @ 1.2 amps (Triad F-14X or equiv.)
V_1—6AK5 tube (triode-connected, see text)
V_2—6J6 tube

Fig. 1. Complete circuit diagram of the FM tuner booster. Circuit within dashed lines is built into one chassis while power-supply components are in another chassis.

"Opened up" view of the two chassis making up the booster. The bottom of the r.f. section is covered for good shielding. Two openings (covered with snap-in plugs) permit access to C_1 and C_4. Layout of power-supply parts is clearly shown at right.



in order for the booster to cover the entire FM band of 20 megacycles it must be tunable since it can only cover 5 megacycles at a time.

This result was to be expected. Since we are trying for high gain, the maximum achievable bandwidth must suffer. This is not a drawback, particularly if only one, or a series of stations located within 5 megacycles of one another, are to be boosted. In this case the booster can be initially tuned to these stations and forgotten. This is so because beyond its 5-megacycle bandwidth the gain of the booster drops, but it must drop 20 db before it attenuates other signals since it has a maximum gain of 20 db. This point does not occur until over 15 megacycles from the center of the booster passband. That is, if the booster is tuned to boost a station at 93 megacycles, then all stations within the 5-megacycle bandwidth of the booster (90.5 to 95.5 megacycles) will be boosted by 20 db. As we move higher in frequency, the gain decreases until it reaches unity 15 megacycles from 93 megacycles, or at 108 megacycles. At this point, the booster provides no gain, but it provides some gain at all other frequencies, rising to a maximum of 20 db at 93 megacycles.

If, on the other hand, it is desirable to be able to boost *any* signal frequency, then the booster must be tuned whenever a station needs boosting.

Since these calculations show that a cascode FM booster can be built with more than adequate gain, such a booster was constructed and is diagrammed in Fig. 1. In order to keep circuit costs low, a simple version of the cascode circuit using direct coupling was used. This minimizes the number of components, particularly coils and variable capacitors. A simple power supply was also adopted to avoid the expense of a power transformer. Total cost of the booster is only about fifteen dollars.

Performance of the cascode booster is excellent. It provides over 20 db gain which has been found more than adequate to pull in FM stations which couldn't be received previously without the booster. Since the author was mainly interested in boosting two stations on the low end of the FM band, trimmers C_1 and C_4 were used to tune the input and output coils and then left.

Constructing the Booster

Since the circuit of the booster is simple, its construction should present no problems. As can be seen in the photographs, the author used two 5" x 7" x 2" chassis screwed back-to-back. The top chassis holds the booster proper and the bottom chassis houses the simple power supply.

Since the booster must amplify signals of high frequency, it is vital that all leads in the r.f. section be kept as short as possible. The shorter the leads, the higher will be the over-all gain of the booster. Tuning capacitors C_1 and C_4 can be either trimmer types if the tuning is to be set but once and forgotten, or variables if the booster is to be tuned to each individual station.

For proper performance of the 6J6, its socket should be "star connected." This merely means that pins 1, 3, and 6 of the socket should be soldered to the center lug of the socket and this center lug grounded with as short a wire as possible. See Fig. 1.

A word of caution; note that the a.c.-d.c. type of "B+" power supply connects the chassis to one side of the power line, making the chassis "hot." This means that the booster should be treated with normal care to prevent shocks. The output of the booster is isolated from ground, therefore your FM tuner will not be connected to the power line *via* the booster. If you wish, a ten-watt or greater isolation transformer can be used to remove all possibility of a shock.

For single station use, the booster must be tuned to this station. If several stations are to be boosted, pick the station closest to the center of the band to be boosted. Tune your FM tuner to this station. Alternately tune capacitors C_1 and C_4 for maximum strong, noise-free reception. The booster is now tuned. If the booster is to be used for all stations, merely tune C_1 and C_4 for best reception.

STARVED AMPLIFIER

BY A. V. J. MARTIN

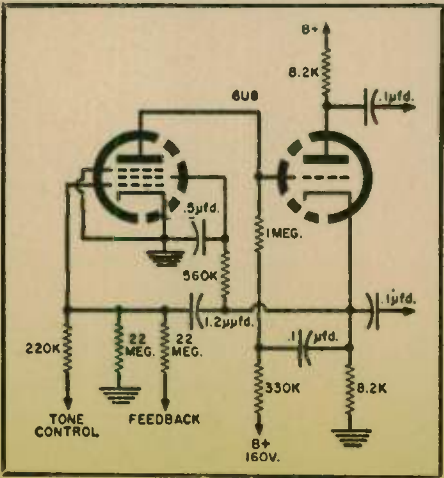
THE starved amplifier circuit, where a pentode is fed very low plate and screen voltages, is interesting because it can provide a very high gain in a single stage.

A 6U8 triode-pentode is used. The input stage is the pentode. The grid circuit contains a tone control and a feedback circuit (not shown). The plate is fed from "B+" of 160 volts through a large value resistor and is directly tied to the grid of the triode, used as a split-load driver for an output push-pull of 6AQ5.

The screen-grid of the input pentode is fed from the cathode voltage of the triode through a decoupling circuit. The triode, acting partially as a d.c. cathode follower, has a stabilizing effect on the input tube.

At high frequencies, an extra amount of stabilization is provided by a small capacitor between the cathode of the triode and the input grid. This technique is being used in a moderately priced audio amplifier of French manufacture.

Circuit gives high gain in single stage.



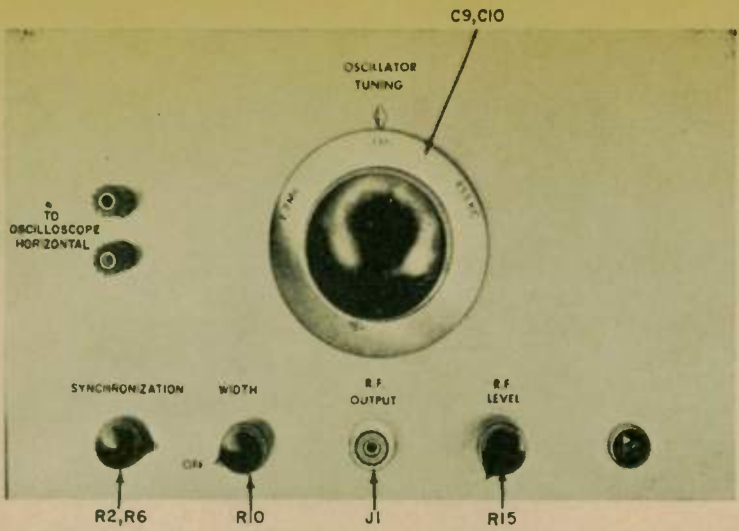


Fig. 1. This simple unit provides the same facilities found in more sophisticated TV-FM sweepers.

A Sweep Generator for Hi-Fi AM

By **DON STONER**

For broad-band audio on AM, sweep alignment is a virtual must—and you can build the right generator.

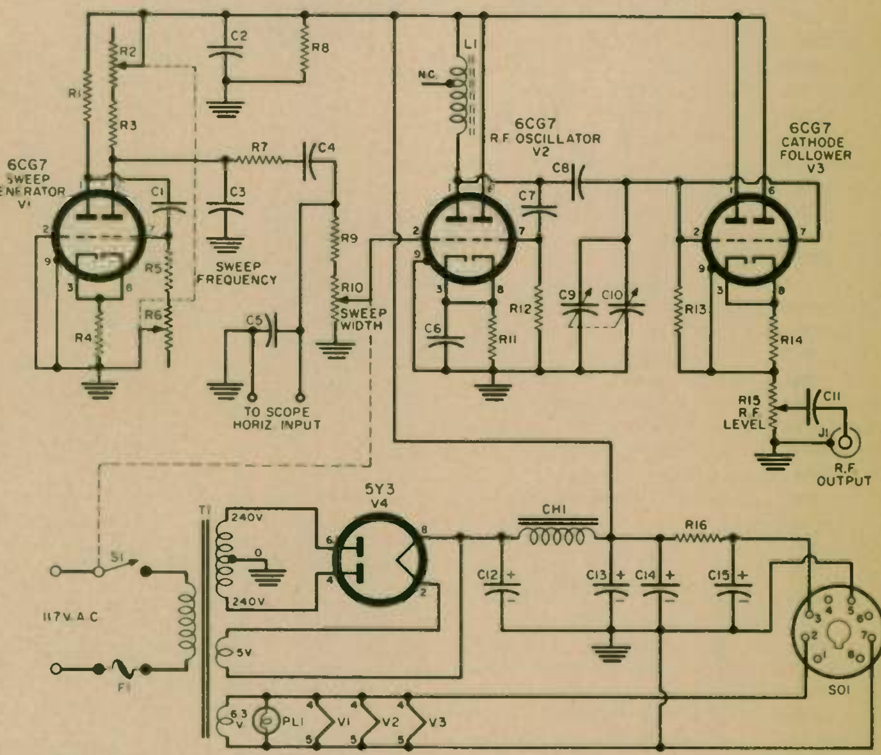
SINCE the popularity of FM broadcasting and reception began to rise rapidly over the past few years, AM stations have come to play "second fiddle" to most people interested in high-fidelity results. This attitude is reflected by the design of some combination AM-FM tuners. Even when great pains have been taken to safeguard the quality of the FM section, the AM portion may consist of nothing more than a ferrite-loop antenna, a converter, an i.f. stage, and a detector.

While detailed alignment instructions for FM may appear, the technician is generally instructed to connect a v.t.v.m. across the a.v.c. load resistor for AM alignment and to peak i.f. transformers and other adjustments for maximum reading. Performance leaves much to be desired, as a rule, and this is due, at least in part, to the alignment procedure.

Although many people still believe that there is an "automatic" limit of about 5000 cps as the highest audio frequency broadcast on AM, there are many transmitters that go much higher. So-called "clear channel" stations may transmit as high as 15,000 cps, the top limit in FM. Others may not go as high, but may transmit up to 10 kc. This is frequently true of AM stations affiliated with FM transmitters that devote some attention to the broadcasting of good music.

The increased interest in stereo broadcasting has made the need for

Fig. 2. Circuit of frequency sweeper, r.f. oscillator, and output stage.



R_1, R—47,000 ohm, 1/2 w. res.
R-R—250,000 ohm/1 megohm ganged pot (Centralab F1-51 and R2-34)
R, R_{11}—470 ohm, 1/2 w. res.
R—27,000 ohm, 1/2 w. res.
R—470,000 ohm, 1/2 w. res.
R—100,000 ohm, 2 w. res.
R—100,000 ohm, 1/2 w. res.
R—1 megohm, linear taper, pot ("Sweep Width," with switch S_1)
R—220,000 ohm, 1/2 w. res.
R—1 megohm, 1/2 w. res.
R—1000 ohm, 1 w. res.
R—10,000 ohm, linear taper, pot ("R. F. Level")
R—3900 ohm, 2 w. res.
C—.22 µf. paper capacitor
C, C, C, C_{11}—.01 µf. mica capacitor
C, C—2 µf., 200 v. paper capacitor
C—680 µµf. silver mica capacitor
C—.001 µf. silver mica capacitor
C-C_{10}—Dual 365 µµf. var. capacitor (Miller #2112)
C-C-C-C—20/20/20/20 µf., 450 v. elec. capacitor (Cornell-Dubilier UPT-222245)
CH_1—7 hy., 50 ma. filter choke (Stancor C-1277)
L_1—Ferrite rod antenna (Miller #2004)
J_1—U.h.f.-type coaxial connector (Amphenol SO-239)
SO_1—8-pin tube socket (Amphenol 168-015)
S_1—S.p.s.t. switch ("On-Off," part of R_{10})
T_1—Power trans. 240-0-240 v. @ 55 ma.; 5 v. @ 2 amps.; 6.3 v. @ 2 amps. (Stancor PM8402 or equiv.)
PL_1—#47 pilot light
F_1—2 amp. fuse
V_1, V_2, V_3—6CG7 tube
V—5Y3 tube

Fig. 3. Accessory probe for the sweep oscillator is used to detect response curve for scope display when alignment or troubleshooting of an individual stage is involved. This unit takes the place of the detector in the AM receiver itself, which performs the same function when an over-all alignment procedure is performed.

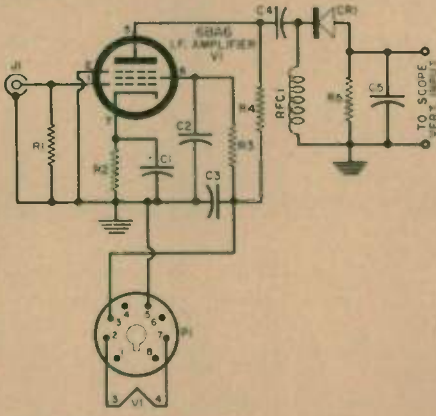
better quality in AM and the means of aligning to obtain that quality more pressing. In most stereo transmissions, the FM station is used for one channel and the other channel is sent out on the affiliated AM transmitter. The necessary balance between the two channels is destroyed when frequency response in the AM receiver is drastically reduced and distortion is increased by peak alignment. To understand the importance of alignment, particularly of the i.f. section, let's review the make-up of the AM signal briefly.

The modulating audio signal is applied to an r.f. carrier that is constant in frequency and that is also constant in amplitude when no audio signal is applied. The modulating signal does nothing to the frequency of this carrier, but it causes carrier amplitude to rise and fall as the audio itself goes positive or negative. Presto! Amplitude modulation.

However, something else of importance takes place. When modulation occurs, although the carrier frequency itself does not change, additional signals at other nearby frequencies are produced. For example, when an audio modulating tone of 1 kc. is applied, two new r.f. signals, or sidebands, are created. These appear on each side of the carrier and are displaced from it by the same number of cycles or kilocycles as the original modulating frequency. In other words, if a station carrier at 640 kc. is modulated by a 1-kc. tone, additional r.f. signals at 639 and 641 kc. would appear. If the highest modulation frequency is 15 kc., then sidebands will be generated as high as 15 kc. above the carrier and as low as 15 kc. below it. The total transmission bandwidth is then 30 kc. wide.

In order to reproduce this modulation faithfully, the bandwidth of the AM receiver (particularly of the i.f. amplifier) must also be 30 kc. wide. If the i.f. stage is no more than 10 kc. wide, as may be the case, audio frequencies above 5000 cps will be lost.

Although it is not always practical, depending on the design of the AM receiver, to realign the i.f. section for the entire bandwidth of 30 kc., it is almost always possible to achieve a worthwhile improvement at some sacrifice in gain.



R_1—2.2 megohm, ½ w. res.
R_2—220 ohm, ½ w. res.
R_3—47,000 ohm, ½ w. res.
R_4—4700 ohm, ½ w. res.
R_5—100,000 ohm, ½ w. res.
C_1, C_3, C_4, C_5—.01 µf. mica or paper capacitor
C_2—.005 µf., 200 v. capacitor
RFC_1—2.5 millihenrys (Miller #6302 or equiv.)
CR_1—1N34A germanium diode
P_1—8-pin octal plug (Amphenol 86-PM8)
J_1—Coaxial connector (Amphenol SO-239)
V_1—6BA6 tube

Fig. 4. Circuit of accessory detector.

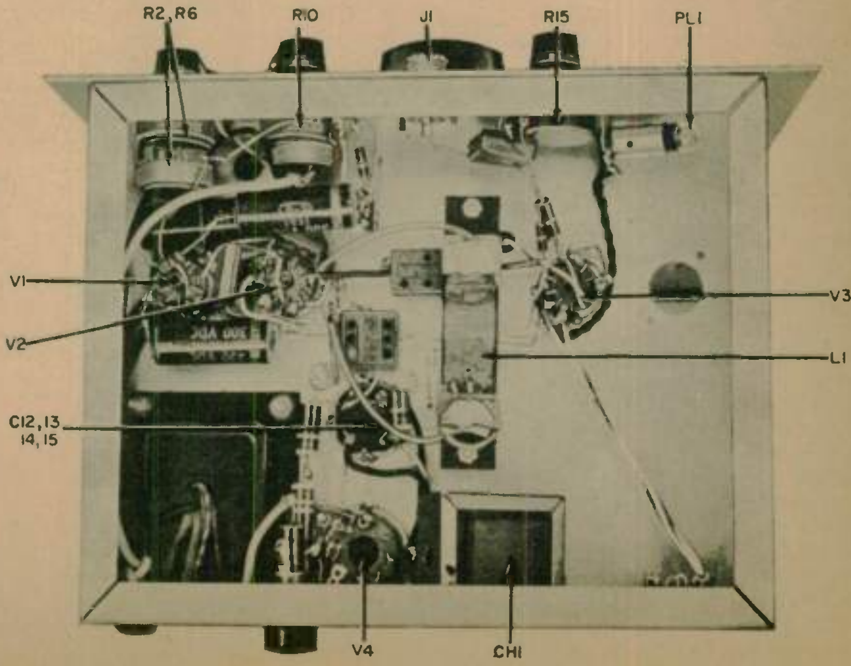
This can be done by stagger tuning; *i.e.*, one i.f. adjustment is detuned slightly from the center frequency in one direction, the next adjustment is tuned off center in the opposite direction, and so on. With an accurate signal generator and a v.t.v.m. as output indicator, the response curve showing the bandwidth can be plotted on graph paper, and replotted as re-adjustments are made.

This process is haphazard and time-consuming. It would be far easier to borrow a technique used on TV receivers, where i.f. bandwidth is an important consideration, and to use a sweep generator while observing the response on an oscilloscope. Unfortunately, there are very few sweep generators available of the type and price that would be practical for a service bench that will operate as low as the frequencies involved in AM reception. The AM sweeper described here (Figs. 1 and 2) was designed to operate in the range between 350 and 1600 kc., which covers all r.f. and most i.f. circuits. In addition, an associated detector probe (Figs. 3 and 4) was designed to assist in troubleshooting, for testing and aligning filters in amateur radio applications, and for general experimentation with tuned circuits.

How It Works

You will note that V_2 (Fig. 2) is employed as a cathode-coupled oscillator. Its frequency is determined by L_1 in conjunction with C_9 and C_{10}. Coil L_1, a ferrite antenna rod designed for use with transistor portable radios, has a "*Q*" of 500 at 790 kc. Capacitors C_9 and C_{10} are the two 365-µµf. sections of a dual, variable unit, with both sections tied together. The total capacity (730 µµf.) allows the coil to resonate as low as 350 kc. To avoid loading the oscillator, the two triodes of another 6CG7 (V_3) are connected in parallel and used

Fig. 5. Underchassis view highlights component layout. Only L_1 may be critical.



as a cathode-follower. A potentiometer in the cathode circuit controls the r.f. output.

A third 6CG7 (V_3) is connected as a cathode-coupled multivibrator. A large capacitor in its output plate circuit, C_3, is charged by the multivibrator output to form a saw-tooth wave. A dual-section potentiometer, R_2 and R_6, controls the speed of this saw-tooth oscillator, which is continuously variable from 5 to 50 cycles. Use of this control will be explained later.

The output of the sweep generator (V_3) appears across potentiometer R_{10} and across the output terminals for the scope. A portion of the signal on R_{10} is fed to the control grid of r.f. oscillator V_2. This signal varies the bias on V_2, and thus causes the frequency to change, at a rate determined by R_2 and R_6 and an amount determined by R_{10}.

The action of V_3 alone is very similar to that of the multivibrator found in the horizontal oscillator of many TV receivers. With R_{10} turned fully counterclockwise (off), the r.f. oscillator (V_2) is not swept and the entire unit may be used as a single-frequency signal generator for regular radio alignment. The saw-tooth appearing across the scope terminals feeds the horizontal input of an oscilloscope for the time-base synchronization. A 5Y3 (V_4) is a full-wave rectifier that supplies power to the sweeper and also to the detector probe.

The probe (Fig. 4) consists of a broadly tuned amplifier and a crystal detector. The detector output is connected to the vertical input of an oscilloscope. This device is very handy when troubleshooting defective i.f. stages, aligning radios or communications receivers one stage at a time, or for experimenting with filters and tuned circuits. It is not necessary to use when aligning hi-fi AM tuners and radios unless they are badly out of alignment.

Construction

The sweeper was constructed on a 7-inch by 9-inch aluminum chassis (Fig. 5). The power section is located in the rear, left-hand corner, directly behind V_1 and V_2. The tuning capacitor (C_9 and C_{10}) stands off the chassis on ½-inch pillars. The extra hole in the chassis next to V_3 was for a voltage-regulator tube. This was subsequently found to be unnecessary and was therefore eliminated. Component layout does not seem to be critical, with the possible exception of coil L_1. This coil should be mounted on ⅞-inch ceramic stand-off insulators to keep it away from the chassis. Naturally the usual precautions of keeping filament wires away from grid connections and components will apply to this unit, to avoid hum pickup. It might be well to mount the filter choke on the top side of the chassis as far away from coil L_1 as possible. The particular choke used by the author did not induce hum in the coil, but others might.

The detector probe layout is entirely straightforward. Since this unit acts as a preamplifier stage for the oscilloscope, it must be connected with shielded cable or with extremely short wires. The operation of this probe may be checked by connecting its output to a scope and connecting an antenna or length of wire to the input coaxial connector. Random jumping traces should be seen on the scope, indicating radio station pickup.

A three-foot length of coax should be used as a sweeper cable. Prepare it by installing a coaxial connector on one end and a pair of clips on the other end. Another three-foot cable with clips on each end connects the saw-tooth output to the horizontal input on the oscilloscope. Also, make two six-inch coax cables for the input and output of the detector probe.

How to Use the Sweeper

To use the sweeper, connect the vertical input of the oscilloscope across the receiver detector load (usually the volume control). Connect the saw-tooth output from the sweeper to the horizontal input of the scope and place the scope function switch on "external sweep input." Connect the sweeper r.f. output (*via* the cable) to the *oscillator* grid of the receiver's mixer tube. Set the oscillator frequency (of the sweeper) to approximately 455 kc. and adjust the r.f. level, sweep width, and synchronization until a swept response-curve trace is observed on the scope. To check the frequency response, loosely couple the output from a regular, accurate signal generator near the mixer. This will put a "birdie" on the trace, similar to a marker used in TV alignment. Grasp the oscillator coil of the AM circuit and observe any changes in the sweep response. If changes are severe, it will be necessary to disconnect the oscillator coil from the mixer tube. In most cases, however, the change will not be significant and it will be just as well to leave the coil in place.

Once a normal i.f. trace is obtained, check the bandwidth at the two points halfway down on either side of the sweep curve. If it is less than 20 kc. wide, stagger the i.f. coil adjustments for the proper bandwidth. It will usually be possible to obtain a satisfactory curve 15 kc. across the top and 20 kc. wide at the half-voltage points. When widening the i.f. amplifier response, you will undoubtedly notice the gain fall off. This is normal whenever the bandwidth is increased.

The sweeper is also useful for troubleshooting of AM radios. Intermittent stages may be isolated by feeding the sweeper into individual stages and connecting the detector probe to the output of each stage. A jumping trace or no trace will quickly pin down the offending stage. Sudden peaks appearing in the response curve indicate regeneration of one or more stages. In addition, the sweeper is extremely valuable for the alignment of multi-stage communications receivers or single-sideband equipment. In fact, it is difficult to do a top-notch alignment job on crystal filter stages without such a sweeper.

The purpose of the synchronization control (R_2 and R_6) may not be clear. Actually, it controls the frequency of the free-running saw-tooth generator. The setting of this adjustment represents a compromise and depends on the particular oscilloscope you use. Generally, the lower the sweep speed, the more accurate the response curve. At five cycles, the response curve represents a very low-frequency square wave and many scopes do not have sufficient low-frequency response to reproduce it. Of course, if you own a scope with d.c.-coupled amplifiers, there is no problem. As the sweep frequency is reduced (counterclockwise rotation), you will notice that the beginning and the end of the response curve shift vertically with respect to each other. The proper setting for the sync control is just before this distortion becomes objectionable. Since the scope time base is derived from the sweeper, there is no particular synchronization problem.

This sweeper can also be very useful to the ham with the particular alignment problems involved in single-sideband operation. Although space limitations prevent covering such use here, it is discussed in various texts on single-sideband techniques.

AUTOMATIC TUNER CLEANING

By KEN RAINS

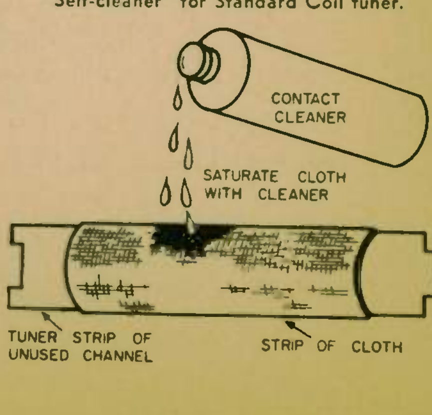
EVERY so often, it becomes necessary to clean the contacts of most Standard Coil tuners to maintain satisfactory performance. This periodic task can become quite annoying, but it can be avoided with a simple measure.

To keep the tuner contacts clean, simply wrap a small strip of soft cloth around the coil strip (or strips) for any one unused channel, as shown, and replace the strip or strips in their original position. Then saturate the cloth with contact cleaner. Whereas such a cleaner, as normally applied, will dissipate fairly rapidly, the cloth acts as a wick, retaining the cleaner for longer periods.

No special measures need be taken to bring about the cleaning action. Rotation of the tuner drum during normal use will clean away grease and dirt, keeping the contacts continuously clean.

—30—

"Self-cleaner" for Standard Coil tuner.



MAIL THE POSTPAID CARD OPPOSITE THIS PAGE AND ENJOY MONEY-SAVING REDUCED RATES ON 1, 2, OR ALL 3 OF THESE WORLD-FAMOUS MAGAZINES!

SPECIAL SHORT-TERM INTRODUCTORY OFFERS — LIMITED TIME ONLY!

HI FI/STEREO REVIEW

9 *months only* **$2.67**

(The single-copy price is 50¢ a month. By comparison, this offer saves you $1.83—the equivalent of more than 3 issues FREE!)

POPULAR ELECTRONICS

8 *months only* **$2**

(The regular subscription rate is 12 months for $4. But this offer gives you 8 months for the price of 6—the equivalent of 2 issues FREE!)

ELECTRONICS WORLD

9 *months only* **$2.67**

(The single-copy price is 50¢ a month. By comparison, this offer saves you $1.83—the equivalent of more than 3 issues FREE!)

These LONG-TERM OFFERS bring you EXTRA SAVINGS!

HI FI/STEREO REVIEW: 36 *months only* **$8.95**

(Compared to the regular $5 a year rate, this offer saves you $6.05!)

POPULAR ELECTRONICS: 28 *months only* **$6.50**

(Compared to the regular $4 a year rate, this offer saves you $2.83!)

ELECTRONICS WORLD: 36 *months only* **$8.95**

(Compared to the regular $5 a year rate, this offer saves you $6.05!)

Take Your Choice ...of any or all three magazines, at reduced rates! ...of the low Short-Term offers or the Long-Term offers that save you even more!

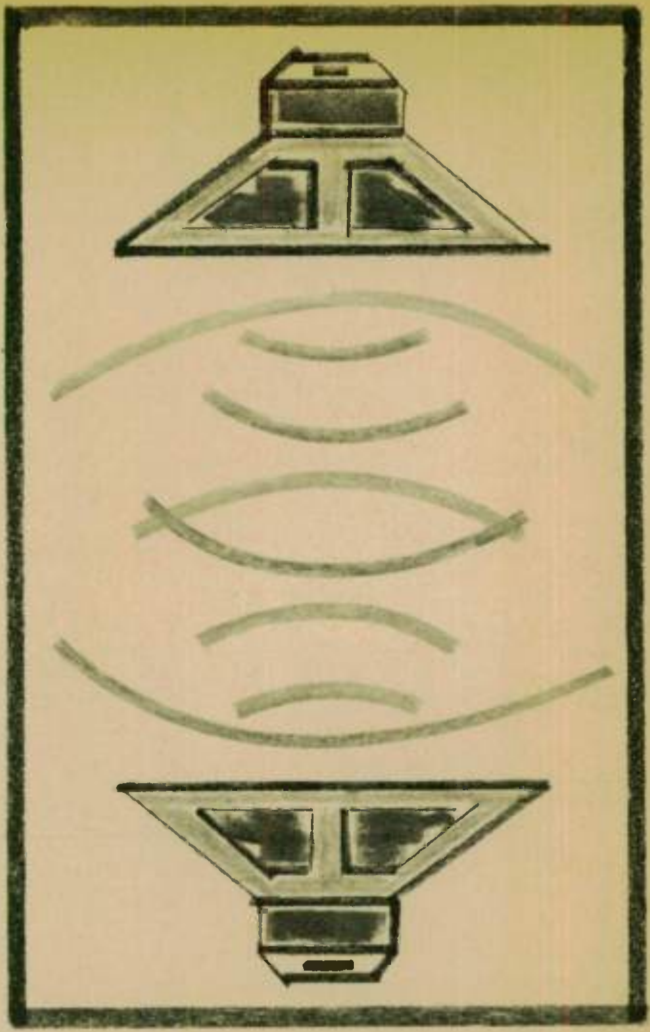
IF SOMEONE HAS ALREADY USED THE POSTPAID CARD...

send your order to: Ziff-Davis Publishing Co., Dept. AY-61, 434 So. Wabash Ave., Chicago 5, Ill.

EXTRA BONUS!

If you enclose payment with your order, saving us billing costs, we will add **1 extra issue FREE** to each Short-Term offer you choose...**2 extra issues FREE** to each Long-Term offer you select!

ZIFF-DAVIS PUBLISHING COMPANY • 434 So. Wabash Ave. • Chicago 5, Illinois



SECTION · 4

Room Acoustics for Stereo

Part 1 – Basic Principles	90
Part 2 – Reverberation and Room Treatment	94
Part 3 – Sound Absorbers and Special Speakers	99
Part 4 – Balancing The Room	104

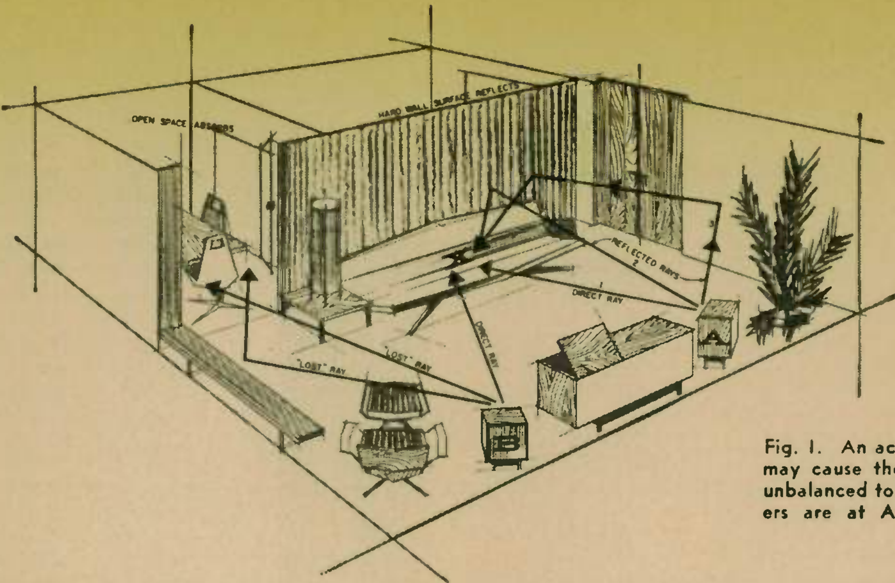


Fig. 1. An acoustically unbalanced room may cause the speaker system to sound unbalanced to the listener. Stereo speakers are at A and B, listener is at X.

Room Acoustics for Stereo

PART 1 BASIC PRINCIPLES

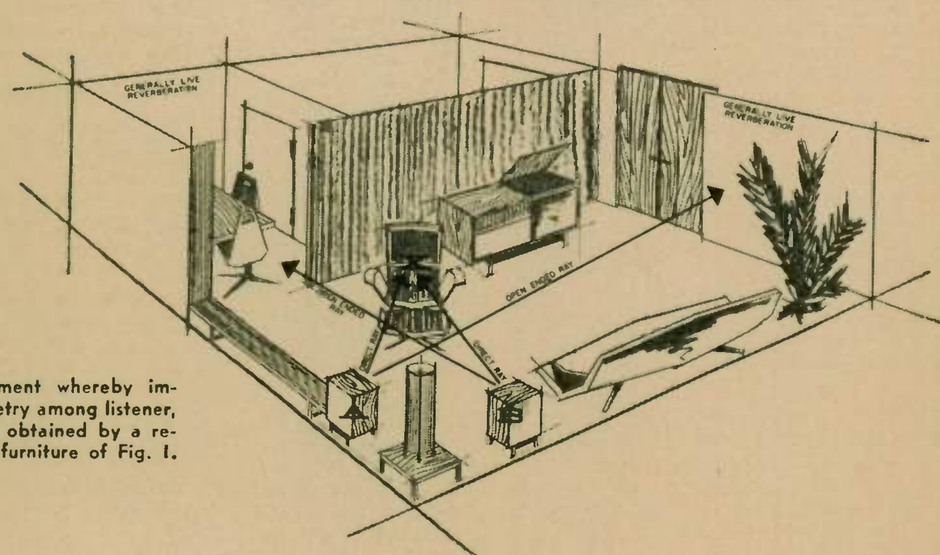


Fig. 2. An arrangement whereby improved acoustic symmetry among listener, speakers, and room is obtained by a rearrangement of room furniture of Fig. 1.

By **ABRAHAM B. COHEN** / Advanced Acoustics Corp.

There is an intimate relation between the stereo speaker system and the listening room. Here are the general principles to be followed for best results.

GOOD sound reproduction requires controlled acoustics in the listening room. For stereo listening, the room acoustic problem becomes even more critical because the reverberant room conditions and reflective wall surfaces may either greatly enhance the stereo effect or completely destroy it.

In achieving optimum stereo effect, there is an intimate relationship between the type of stereo speaker system used and the acoustic conditions of the room. Because of the many different types of speaker systems and the innumerable types of listening rooms, a simplified approach to the speaker-room combination will be covered. This approach may then be extended to diverse types of loudspeaker systems.

Proper speaker placement for high-fidelity stereo reproduction is considerably more complex than for high-fidelity monophonic program material. In single-channel reproduction we have but one problem as far as the loudspeaker is concerned, *i.e.*, at which spot in the room will the loudspeaker perform best? With stereo we have a compound problem which involves: (*a*) how far apart shall the speakers be? and (*b*) how shall they be angled toward the listening area? This leads to the next point (*c*) which involves the best place to sit, stand, or recline to get the best stereo effect. Unfortunately, many articles in the stereo literature treat only lightly one additional factor that influences all of these considerations. This factor is the effect of the acoustical condition of the room in which the "stereorama" is displayed.

Psycho-Acoustic Enlargement

The conditions which characterize a good listening room, whether for monophonic or stereo service, involve two simple factors. The first is that the room be reverberant to the proper degree to give the reproduced sound "liveness." The room should not be too reverberant for then the reproduced sound would appear too cavernous, hollow, and indistinct. Nor should the room be overstuffed for then it would sound dead and somber. To a considerable degree, stereo systems themselves produce a psycho-acoustical revision of the absolute acoustic properties of the listening room. This, in effect, may make a room seem larger than it really is simply because of the apparent discrete localization of different sections of a large orchestra to separated areas within the room. The very essence of stereo sound reproduction thus "pushes" the walls of the room apart. Our minds automatically tend to convert this "spread out" reproduced orchestra into a "stage" for the orchestra.

Such psycho-acoustic enlargement, when listening to monophonic, or single-channel, program material radiated from one central speaker opening, would be virtually impossible. Some efforts are made to produce artificial enlargement of monochannel recordings at the recording studio by the judicious injection of controlled amounts of reverberation. There have been many "pop" selections recorded with this artificial reverberation added — but in many instances so poorly done that the music and singer are all but lost in the muddled "barrel of sound."

Thus, in monochannel reproduction, size of the room in "undirectional," that is, controlled in a "reverberation direction" but only by the reverberation inserted at the source. This is in contrast to stereo recording where spaciousness is not only injected at the source but is, moreover, under the control of the listener with respect to placement of the speakers and their acoustical environment.

The fact that there has been a fad for injecting reverberation into monophonic recordings, even though overdone, is indicative of the feeling that "spaciousness" does add psychological "depth" to our listening area. There are even electrical reverberation devices, intended for home use, which are designed to "liven up" (psycho-acoustically) the monochannel reproduction. In the case of monophonic reproduction, however, where the original source of the reproduced sound is a single cabinet, it is more difficult to get a wide-stage psycho-acoustic effect without resorting to these highly accentuated effects incorporated in the original selection. It would be considerably better to make the recording under natural reverberation conditions and let the acoustic enlargement be obtained by means of the listening room environment. This, however, is a rather utopian ideal. Pehaps someday the standard living room will come with adjustable sound-conditioning facilities similar to the air-conditioning systems now in widespread use.

In the early days of stereophony, it was frequently noted that, when a switch was made from monophonic to stereophonic playback, to get the same "loudness" effect, the total gain of the amplifiers had to be reduced. Perhaps these were the first straws in the (acoustic) wind that tended to show that with monophonic reproduction the ear needed more re-enforcement from its surroundings to get the same loudness effect as for the stereo reproduction. From this we may possibly conclude that a "live-r" reverberation condition is desirable for monophonic reproduction than for stereo reproduction. This leads us to the thought that it would be desirable not only to be able to correct our listening rooms in general so as to present the proper acoustic environment but to make this acoustic adjustment variable to accommodate changing conditions of playback methods, types of programs, or number of people in the room.

Stereo has the latent power to really put us in the concert hall, spatially, without monophonically overdone artificial effects if the speaker system is properly balanced in performance and is judiciously installed in a manner consistent with *good concert hall practice*. This means that we must look to our living room for the last link in the "realism" chain, with the aim of bringing it as close to good musical acoustic practice as possible.

Reverberation Adjustment

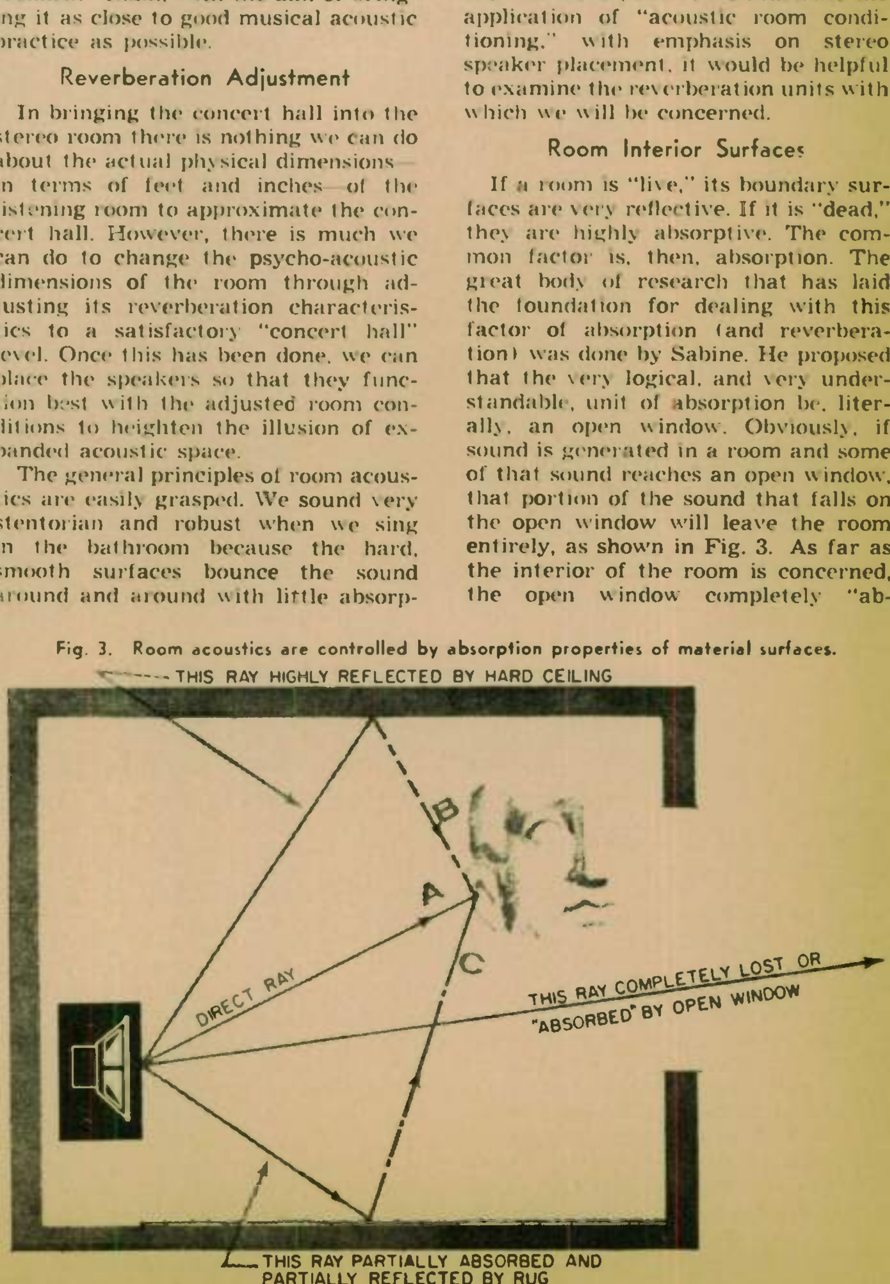
In bringing the concert hall into the stereo room there is nothing we can do about the actual physical dimensions — in terms of feet and inches — of the listening room to approximate the concert hall. However, there is much we can do to change the psycho-acoustic dimensions of the room through adjusting its reverberation characteristics to a satisfactory "concert hall" level. Once this has been done, we can place the speakers so that they function best with the adjusted room conditions to heighten the illusion of expanded acoustic space.

The general principles of room acoustics are easily grasped. We sound very stentorian and robust when we sing in the bathroom because the hard, smooth surfaces bounce the sound around and around with little absorption so that the sound seems to persist loud (but not necessarily clear) as we continue to bellow to our heart's content. On the other hand, if we were enmeshed deep in a well-stocked clothes closet, we could shout until we turned blue with exertion and would still sound weak and feeble. The hard reflective surfaces of the bathroom make the room "live." The soft, absorbent "stuffing" of the clothes closet makes the closet "dead." In between these acoustical extremes lies an acceptable mean for good concert hall — or for that matter — listening room liveness. Our acoustic goal is to achieve a room condition which is live enough to give fullness and body to all the musical resonances yet not so live that recurrent echoes will blur the original rich sound. There have been many determinations as to the optimum degree of liveness for the concert hall, and these same principles may be readily applied to your own stereo listening area. However, before we illustrate the application of "acoustic room conditioning," with emphasis on stereo speaker placement, it would be helpful to examine the reverberation units with which we will be concerned.

Room Interior Surfaces

If a room is "live," its boundary surfaces are very reflective. If it is "dead," they are highly absorptive. The common factor is, then, absorption. The great body of research that has laid the foundation for dealing with this factor of absorption (and reverberation) was done by Sabine. He proposed that the very logical, and very understandable, unit of absorption be, literally, an open window. Obviously, if sound is generated in a room and some of that sound reaches an open window, that portion of the sound that falls on the open window will leave the room entirely, as shown in Fig. 3. As far as the interior of the room is concerned, the open window completely "ab-

Fig. 3. Room acoustics are controlled by absorption properties of material surfaces.



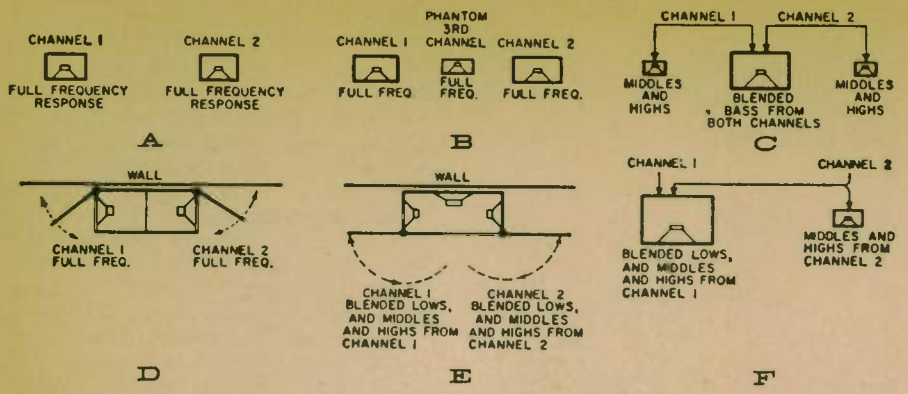


Fig. 4. There are many stereo speaker systems in use today as is shown above.

sorbed" the sound that hit it—for none of that sound found its way back into the room. A square foot of open space in a wall thus becomes an "absorption unit."

Nothing, but nothing, can quite equal the absorption of a hole in a wall looking out into the world. Whatever the material may be, if it is physical, if it can be weighed, calipered, touched—it *will absorb some sound, reflect some sound, and transmit some sound.* Substantive materials thus have individualistic absorption characteristics defined by their "absorption coefficient" in relation to "open space." Whatever the material being considered, its absorption is always less than "open space," a square foot of which is considered to have "unit absorption." Thus, the absorption coefficient of any material is always less than 1.

Referring to Fig. 3, the sound from the speaker system that finds its way out of the window will be completely lost as far as further usefulness to the room is concerned. It has, in effect, all been absorbed by the open window. On the other hand, the sound from the loudspeaker which hits the carpet is partly absorbed and partly reflected back into the room. Likewise, the sound hitting the plastered walls and ceiling is absorbed to a lesser degree and is reflected back into the room with more intensity than is the case with sound striking the carpet. As far as the listener is concerned, the aggregate sound that reaches him is composed of the direct ray *A*, the highly reflected ray *B*, and the somewhat reflected ray *C*. It is obvious, then, that the degree of liveness, or "concert-hall-ism," presented to the listener by his room is controlled by the absorptive properties of his surroundings.

Special Stereo Needs

This problem of the room condition is important for any type of sound reproduction but for stereo it is doubly important (no pun intended). For the sake of preliminary illustration of the effect of the acoustics of a room on the resultant sound field of a stereo speaker system, we will use as the stereo sound source two speakers, 6 to 8 feet apart, oriented toward the central listening area by approximately 15 degrees each and we will assume that the speakers are as nearly balanced in their performance as is possible.

Now, even though we accept this "paired speaker" condition as being most suitable for our illustration, it should not be inferred that it is the only acceptable stereo speaker configuration. Much definitive work still needs to be done on speaker pairs (or trios), balanced or unbalanced, and we shall have considerable comment to offer concerning this problem after we have explored the preliminary example of the effect of room acoustics on the reproduced sound from the balanced and separated speaker system.

Suppose, now, that we place our perfectly balanced speakers, at the previously specified distance and angular relationships, in a room typical of many new home constructions where the living room and the dining area form a sort of an "L," as shown in Fig. 1. The speakers are neatly balanced on either side of the equipment cabinet. At the opposite wall is a foam rubber settee backed up against a hard plasterboard wall (decoratively papered, of course) with probably an arrangement of picture miniatures hung above the couch (a large, heavy picture would probably rip its supporting hooks right out of the plasterboard). In any event, small pictures or large, the backing wall will be rather "hard" acoustically—it will reflect quite a bit of the sound that hits it.

We turn on our stereo system, lower ourselves onto the comfortable settee and listen. But although our reproducing speakers are balanced, phased, separated, and angled properly, the *listening system* is no longer balanced. Not even the channel-balancing control on the amplifiers or "stereo control center" will correct this acoustically unbalanced room.

Speaker "A" sends a direct sound to the listener *via* ray 1 and considerable wall-reflected sound to the listener *via* rays 2 and 3. On the other hand, Speaker "B," while it may transmit direct sound to the listener, sends very little reflected sound to his ears for there is the "open space" of the dining ell in front of it. Thus the sound from Speaker "A" will be louder *to the listener.* A second effect is that there will be considerable high-frequency loss from Speaker "B" *as heard* by the listener. The high frequencies from Speaker "B" that continue on straight ahead are almost completely lost to the listener while the highs from Speaker "A" come to him directly *and* by reflection. Consequently, although the speakers may have been perfectly balanced acoustically, their reproduction at the ear's location is unbalanced, with the unbalance being preponderantly in the high-frequency range, due to the differential reflective conditions bridging two speakers and the listener. This is unfortunate because much of the stereo effect is contributed by the higher frequencies. If we lose the highs, we lose the directional sense of the system.

We may overcome these deleterious effects of the distortion of the direction of the sound by one of two methods —or preferably—a combination of both. The first obvious method is to re-arrange the furniture; the second method is to alter the reflective properties of the room. Either of these problems may prove difficult to overcome unless one finds some way of convincing the lady of the house that it is actually *her* idea to re-arrange the furniture, or to hang a drape here or there, or to install acoustic tiles hither and yon. Having thus easily (!) surmounted this first "managerial" problem, let us see what might be done with the furniture arrangement shown in the room of Fig. 1 to provide a better acoustic balance among the speakers, room, and the listener.

Fig. 2 shows one alternate arrangement of the speakers where *almost* complete acoustic symmetry among the speakers, room, and listener is available. The listener is in the direct way of each speaker. Each speaker also faces almost equivalent open-ended spaces, that is, Speaker "B" radiates into the dining ell and Speaker "A" radiates into the living room. To the listener, then, there is presented a condition wherein there is a balance between the direct rays from each speaker and the reflected rays.

There is also a second-order effect, but a desirable one. If the rooms are moderately "live" and the program is played at good volume (who ever heard of low volume?)—then an additional acoustic enlargement of the room takes place. The sound components from the speakers that travel into the dining ell area and into the living-room area find themselves bouncing around within these somewhat live end spaces. A "liveness" effect—or reverberance—is thus produced which adds to the direct rays received at the listening area. This partly gives rise to a psycho-acoustical enlargement of the listening room in a way that somewhat simulates the larger chamber music room—provided, of course, that the sound is reproduced loud enough to energize these reflective areas. This arrangement of Fig. 2, though quite severely limiting the effective stereo listening area, does provide a solution (for illustrative purpose) of balanced stereo listening.

Some Different Approaches

Before we opened the discussion of the above problem of the acoustics of an unbalanced room upon a balanced speaker system, we deferred our comments about other types of speaker systems and configurations for stereo application. It would now be well to give this matter some consideration before going into the specific problem of room acoustical analysis and adjustment where these different types of speaker systems are used.

There are in use today speaker systems of many types which, as shown in Fig. 4, although greatly different, yield discernible and psycho-acoustically acceptable stereo. In addition to the two separate speaker systems (Fig. 4A), there are systems which consist of two full-range end systems with a "phantom fill" as a third channel in the middle (Fig. 4B). There is a system where the major bass information is all blended in one central system and the higher frequency stereo-determining components are displayed by end-placed outboard speaker components (Fig. 4C). There is also a system which is enclosed in one cabinet where the stereo effect is produced by reflecting into the room—from each of its *back* hinged end doors—the full program content of each channel respectively (Fig. 4D). In addition, there is a system where the lows are blended in one woofer and dispersed by the rear wall, and the high-frequency components of each channel are guided by the *front* hinged end doors to be randomly reflected by the room walls (Fig. 4E). Finally (at this writing, at least), there is the system where the blended lows and highs from one channel radiate from one system and only the highs from the other channel have a separate speaker (Fig. 4F).

Obviously, not all these systems are "best" systems. They each have their individual merits and drawbacks. And obviously, they cannot all produce exact replicas of the original sound coming out of the instruments. But one thing they all do. They give *some* impression of the stereo effect, whether it be a correct or an incorrect one, to which we then add our own personally devised psycho-acoustic image of what we think the stereo reproduction should sound like—and we then have "stereo."

The question then arises how do we treat all these systems when integrating them to the acoustical condition of a particular room. Unfortunately, there is no definite and rigorously circumscribed "best" relation between the speakers of these various systems and any room at random. The number of solutions would obviously approach infinity, depending upon room size, shape, acoustic condition, speaker system, the listening habits of the auditor, and the type of music being played.

To briefly illustrate the complexity of the problem, take the last item just noted—the type of music being reproduced in relation to speaker system and room acoustics. If the stereo program material were that of a string quartet, we would expect, for stereo realism, that the stereo effect encompass a rather small area governed by the usual intimate geometrical relation among the two violins, viola, and cello, as shown in Fig. 5. If the speaker system were an adjustable one in the matter of speaker separation and speaker orientation—as would be possible with the units of Figs. 4A, 4B, 4C, and 4F—then after auditing a few programs the listener might be able to strike an arrangement where the stereo effect produced would be fairly close to the original sound distribution. The arrangement in Fig. 4D could also be acceptably adopted so that the doors, hung from the back edges, and properly angled, would project the quartet image into a centrally confined area with a minimum of deteriorating acoustic splash from the side walls which would otherwise tend to swell the "size" of the quartet. Fig. 4E would probably not be as adaptable to this type of music for, at best, even with the front-hung doors completely closed (onto the front) the stereo-determining middle- and high-frequency components would still be strongly projected towards the side walls of the room and would reach the listener with strong reflected level from these comparatively distant areas with the result that, again, the quartet's geometrical reproduction would be "swelled."

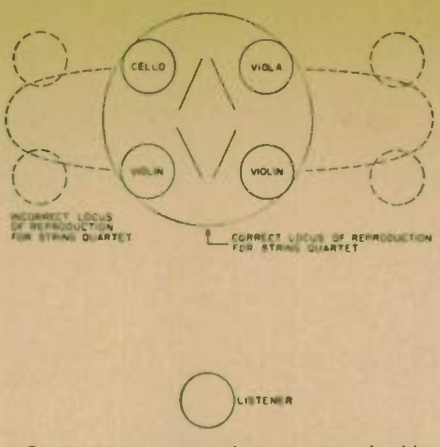
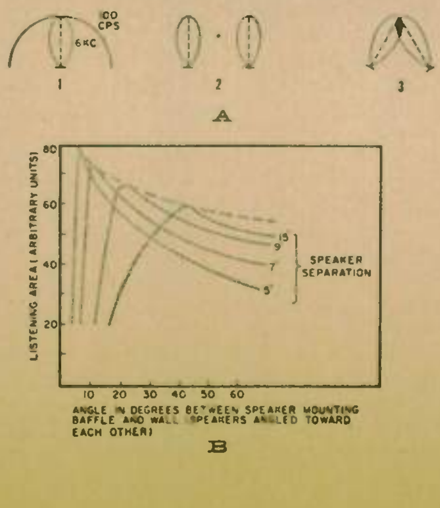


Fig. 5. A stereo reproducing system should be able to "place" the instruments in a manner duplicating their original location.

Fig. 6. (A) Higher frequency radiation should overlap for best results. The overlap area is determined by speaker separation and angular orientation. (B) Graph of maximum stereo listening area as a function of speaker separation and orientation, based on radiation from 12" loudspeakers.



On the other hand, if the reproduced program were a full-bodied and large symphony orchestra, then this last system with its doors wide open, would direct the sound toward the outer reflecting wall to give a resultant enlarged acoustical image more in keeping with the original effect. The same effect could, of course, be obtained with the other systems through greater separation of the individual speakers, controlled center fill, and system angling. Perhaps this brief dissertation on the correlation of the reflecting properties of the room, the speaker system philosophy, and the program material will bring home the difficulty of achieving one perfect solution—obviously a single solution to suit all conditions is out of the question. But, despite the complexity of the problem, or rather because of its complexity, we must make some simplifying assumptions in an effort to approach analytical results which may then be applied to more complex systems.

Balanced Speaker System

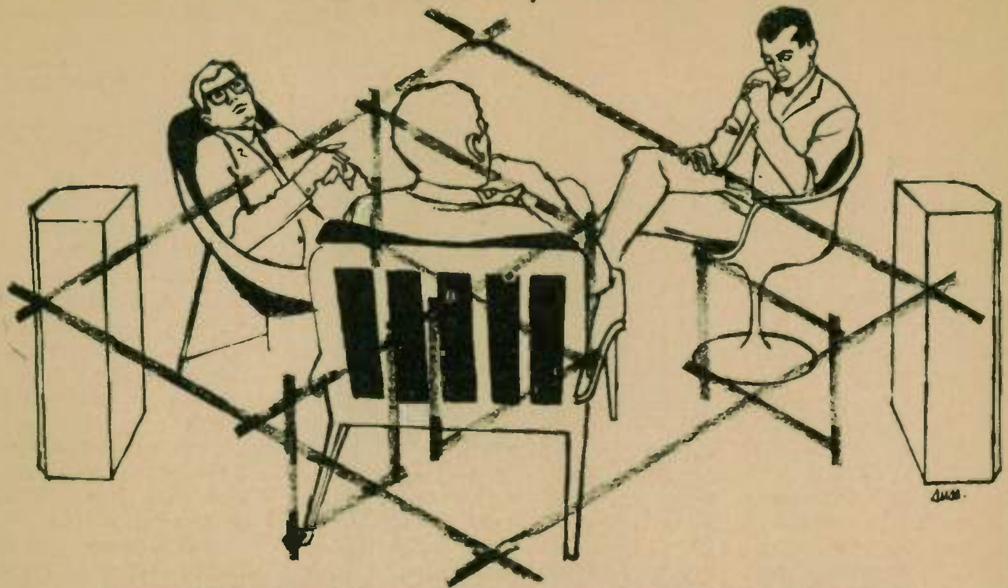
Even if we limit ourselves to analysis of the stereo effect of two identical systems, we may run into an imponderably large number of physical geometrical relationships between the two loudspeakers that do not necessarily bear any definite relation to the room characteristic. In order to resolve this matter of the placement of two identical systems in order to get the maximum stereo effect, the writer undertook to examine the matter in an analytical fashion in a paper which was presented before the Audio Engineering Society Convention in September 1958. The analysis endeavored to determine how far apart two identical speakers should be placed and how they should be angled toward each other in order to obtain the maximum *area* in which the stereo effect would be perceived when the individual speaker system consisted of a simple 12" speaker. One of the results of this analysis is shown in Fig. 6 from which it is seen that *for such a system* the maximum stereo listening area will exist when the speakers are placed about 7 feet apart and are both toed-in by about 15 degrees from the wall against which they are mounted. Now, obviously, this result cannot be accepted as gospel truth for all of the other types of speaker systems previously illustrated but it does at least give us a toe-hold on a premise that will permit us to discuss the acoustics of a room and its adjustment in terms of a frequency space pattern of which we can be reasonably sure.

After we have gone more intensively into the study of the room itself and its acoustics, we may then be able to convert these findings for the balanced system for application to those situations where the composite type of system is to be employed.

Room Acoustics for Stereo

Part 2. Reverberation & Room Treatment

By **ABRAHAM B. COHEN** / Advanced Acoustics Corp.



How the optimum room reverberation is obtained and room resonances are controlled by the use of standard decorative and special acoustic materials.

ONE requirement for a good listening room is that it have the proper reverberation time—or liveliness—to simulate good concert hall conditions. Control of reverberation may be achieved by using standard home decorative materials and commercial sound absorbent tiles. By means of absorption tables covering architectural materials used in home construction, ordinary decorative materials, and commercially available acoustic tiles, it is fairly simple to calculate and "correct" a room to provide proper reverberation conditions.

Reverberation

A room in which sound is to be produced or reproduced, whether live speech and music direct from the piano sounding board or reproduced sound from loudspeakers (monophonic or stereo), must meet certain acoustical conditions if the sound is to be heard properly. A good "listening" room should have correct reverberation properties and generate a minimum of multiple echoes.

Analytically, reverberation is a measure of how fast a room will cause sound to be reduced to a certain level *after* the source of sound has ceased to emit its signal, *providing* the sound has been fully built up in and has "saturated" the room before the source was stopped. We naively state that reverberation is the quality of "liveness" of a room (to sound) but what is the actual mechanism of room acoustics that leads to reverberation?

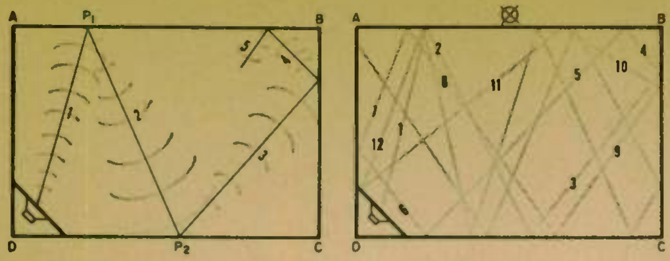
In Fig. 7A let the loudspeaker in the corner generate a constant-frequency and constant-amplitude sound. The wavefront emerging from the loudspeaker builds out in a spherical shell of constantly enlarging radius. As soon as that first wavefront reaches a reflecting surface, as at point P1 on wall A-B, the wavefront will be reflected. This reflection travels outward from the wall just as if there were an additional actual sound source at wall A-B. Again the sound from this new source travels out in ever-increasing spherical fronts until it hits another surface like wall D-C from which it will again be reflected as if there were another sound source at P2 on that wall. Thus these reflections will continue until there are a great number of these reflected rays travelling through the room, as shown in Fig. 7B, producing a generally diffused sound condition throughout the room—*as long as the source continues to operate.*

When the source of sound stops, the diffuse reflections do not stop immediately because the sound that has left the speaker has a finite speed of travel. The sound will keep reflecting in a completely random manner, each reflection being successively weaker than the preceding one. The more absorbent the walls, the less sound power gets reflected and the quicker the sound dies out. When the sound dies out slowly, there is little absorption within the room, and the room would be termed "highly reverberant."

Reverberation, then, is correlated with time—how fast does the sound die out? This is a measure of reverberation—"time." Consequently, "reverberation time" has been defined as the time (in seconds) that a steady-state sound in a room decreases to 1-millionth (1/1,000,000) of its original intensity (after the source has stopped). See Fig. 8. In easier terms, when the intensity of the sound, *after the source has stopped,* decreases by 60 db, the time it took to drop to this level is the reverberation time.

Now, just what sort of reverberation time period should one expect of a listening room, especially one intended for stereo listening? In Part 1 of this series we noted that perhaps a shorter reverberation time for stereo might be desirable since the very essence of stereo itself "psycho-enlarges" the room. This question of optimum reverberation time cannot be answered by complete specifications down to the last decimal place. It is a variable, depending on room or auditorium size and the type of program material being presented. Studies have shown that while an auditorium sounds right—has an accepted reverberation time—for a full symphony orchestra, it may not have the proper reverberation for chamber music or voice.

Many studies have been made of music rooms, auditoriums, concert halls, churches, and similar structures to determine the acceptable range of reverberation times and these figures have been generalized in Fig. 9. While it would be nice to have an instrument at home which would enable us to check the reverberation time of our listening room against this sort of



A B

Fig. 7. When a sound source continues to operate, diffuse reflection builds up until the sound level reaches steady state.

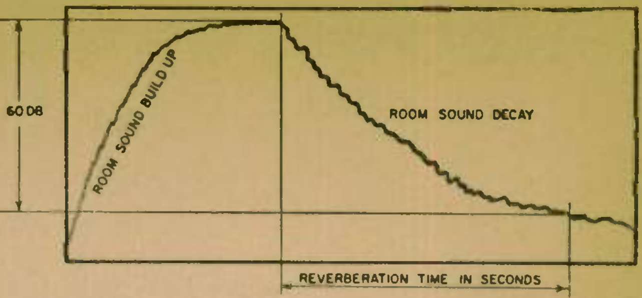


Fig. 8. Reverberation time is the time, in seconds, taken for steady sound to be reduced to a millionth of its original level.

chart, such equipment is quite expensive (for home use) and requires an experienced professional hand at the controls. For such non-critical conditions in the home it is possible to calculate the reverberation time of any room by measuring its physical size and applying information from the charts and tables to be discussed shortly.

In very simple form, $T = .05V/A$, where T is the reverberation time of the room (in seconds), V is the volume of the room (in cubic feet), and A is the *total* absorption of the room and its contents. V, the volume of the room, is no trouble to calculate. Total absorption may be judiciously estimated by reference to established tables which give the absorption coefficients of various materials used in the construction and decoration of the room. For purposes of this article, we must, of necessity, limit ourselves in listing absorption coefficients. What we will do is list the more common household materials which may affect the acoustic conditions of our listening room and then work with these.

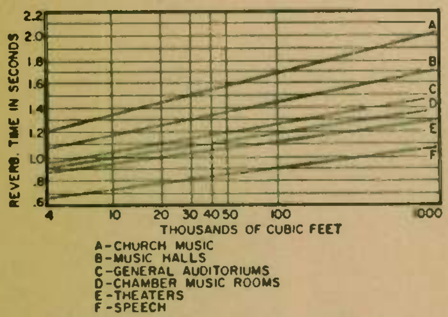


Fig. 9. Optimum reverberation times.

Table 1 itemizes such common household materials while Fig. 10 illustrates, in chart form, the absorption characteristics, as a function of frequency, of some representative commercial materials available for acoustical correction purposes. It will be observed that, in general, the softer the material, the more it soaks up the sound.

Another interesting item to be observed from Table 1 is that a given material can be arranged so that its absorption characteristic may be varied. Velours, for example, when hung flat, have much less absorption *per unit area,* as shown in Fig. 11 than, if we drape or "bunch" a length of velour into an area one-half its flat length. As Table 1 indicates, we practically double the absorption. This, then, is a basic principle by which highly absorbent surfaces are obtained . . . "amplify" a given absorptive area by providing a dimension *in depth* for more absorption.

Acoustical Tiles

Most of us are familiar with those off-white squares of "sound absorbent" material widely used on the ceilings of public buildings. Such materials are known as "tiles" in the trade. Fig. 13 shows some typical units of this type. The tiles are generally a foot square and may be obtained in several thicknesses—¾" being standard for home use. The tiles are made in a variety of materials and textures. Some are simply rough surfaces with "cracks" or fissures apparent on the surface while others are of the perforated variety, some with evenly spaced holes and some with randomly spaced perforations.

Although an inspection of Fig. 10 indicates that the absorption characteristics of these materials are roughly similar, the perforated variety has some advantages in absolute absorption, especially in the mid-frequency range, although the "curve" of absorption is not as smooth as for the unperforated type.

If such acoustic tile is to be applied for the purposes of reducing noise due to the human voice alone, as in restaurants and offices, then the perforated material, with its accentuated mid-frequency absorption (where most of the voice frequencies are found) is perhaps most suitable since the ear is especially sensitive in this area.

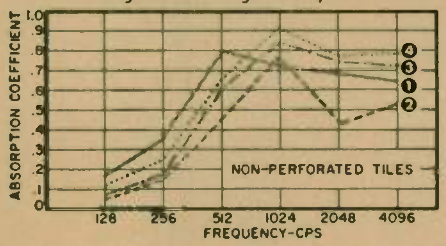
If, however, we wish to absorb a fuller range of sound than that of the human voice, then the fissured material is preferable because of the smoother absorptive curve, although more of the material may be required. Moreover, as far as the home is concerned, the fissured or sculptured material will fit in with the decor better and look less "industrial" than the perforated variety.

It is interesting to note that the rise in the absorption curve of these materials roughly coincides with the frequency range where the stereo differentiating effect takes place. In the region above 300 cps, spatial differentiation becomes effective and separate speakers reproducing these frequencies are necessary for proper spatial perspective. Thus the effectiveness of these sound absorbing materials roughly matches the frequency range where the stereo effect becomes discernible—making them eminently suited for acoustic adjustments and compensation for stereo reproduction.

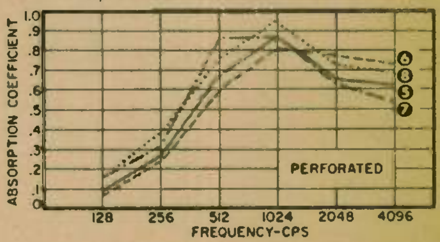
Determining Reverberation

To illustrate the application of the reverberation formulas in conjunction with the tables, let's choose a room with the typical dimensions shown in Fig. 12. In its bare, untenanted state, the room consists of a pine floor with the walls and ceiling of plaster, lime and sand finish on metal lathing. The

Fig. 10. Average absorption characteristics of representative acoustic tiles.

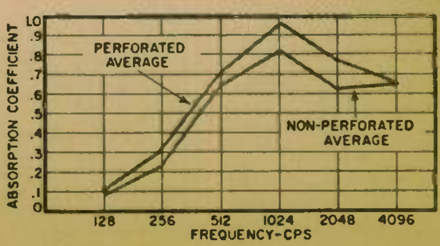


A



B

CURVE	MANUFACTURER	COMPOSITION
1	INSULITE CO.	COMPRESSED WOOD PULP
2	ARMSTRONG CORK	CORK
3	NAT'L GYPSUM	COMPRESSED WOOD FIBERS
4	AMER. ACOUSTICS	CORK GRANS. & MINERAL BINDER
5	JOHNS-MANSFIELD	FELTED MINERAL FIBERS
6	CELOTEX CORP.	MINERAL FIBERS
7	OWENS CORNING	FELTED MINERAL FIBERS
8	ARMSTRONG CORK	CORK COMPOUND



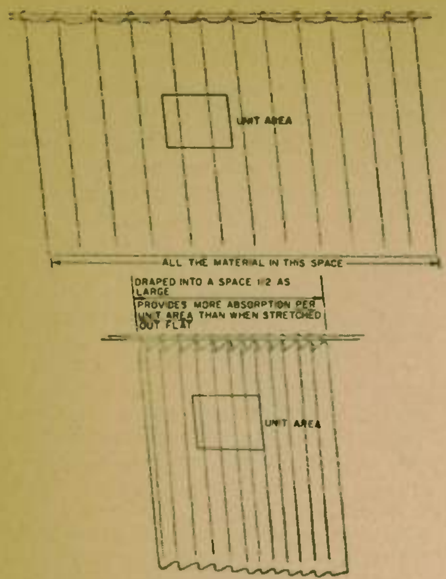
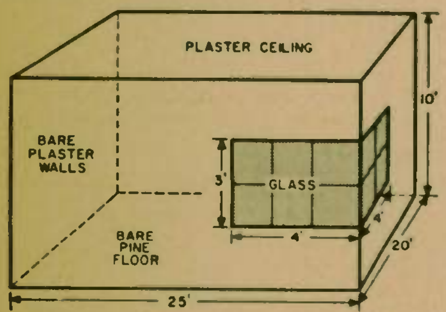


Fig. 11. Here is a simple method of changing the amount of absorption in a typical listening room, as required.

Fig. 12. This completely bare hard room will be too "live", while the room will be too "dead" if completely padded. Figures used are shown below.



	Surface	Sq.Ft	X Abs. Coeff. (512 cps)	= Total Abs.
Completely bare room	Floor	500	.09	45
	Ceiling	500	.06	30
	Walls	876	.06	52.6
	(2 @ 25 X 10 = 500) + (2 @ 20 X 10 = 400) = 900 − glass (2 @ 3 X 4 = 24) = 876			
	Glass	24	.03	0.7
				128.3 units
	T = (.05 X 20 X 25 X 10) / 128.3 = 1.95 seconds			
Completely padded room	Floor (carpeted)	500	.37	185
	Ceiling (bare)	500	.06	30
	Walls (draped & assuming windows covered by drapes)	900	.55	495
				710 units
	T = (.05 X 5000) / 710 = .35 second			
Adjusted room	Floor (carpeted)	500	.37	185
	Ceiling (bare)	500	.06	30
	Walls (draped 1/5 of area)	180	.55	99
	Walls (bare plaster 4/5 of area)	720	.06	43
				357 units
	T = (.05 X 5000) / 357 = .70 second			

windows, of course, are of standard glass. We know intuitively how a loudspeaker would sound in this bare room, but just as a simple exercise to illustrate the use of our "mathematical tools," let us put down some numbers that will provide the figure of (de)merit for this room.

We are going to apply the reverberation formula ($T = .05V/A$) previously discussed, which relates room volume to total absorption. To get total absorption, we simply add up all the *absorption units* of all the surfaces of the room by *multiplying the areas of the various exposed surfaces by their respective absorption coefficients.* In simple cases such as this, the absorption coefficient for 512 cps is used. Thus, as indicated in Fig. 12, the floor has an area of 25′ x 20′ or 500 square feet. It is made of pine flooring which has an absorption coefficient of .09 (per square foot). Its total absorption is 500 x .09 or 45 *units.* The "units" for the walls, ceiling, and windows are figured in the same way on the basis of their areas and absorption coefficients. The absorption of the room totals 128 units and, as indicated in Fig. 12, the reverberation time becomes 1.95 seconds. Checking back to Fig. 9 we find that this reverberation time is far too long for proper definition of sound in a room this size (5000 cubic feet). The room will be too "live" which, of course, we knew from the start.

Now let's go to the other extreme and drape all the walls (using a drape which has an absorption coefficient of .55) and lay carpeting on the floor. The second tabulation, Fig. 12, shows that the total absorption will now be 710 units. Again applying our reverberation formula we get .35 second, which we can see from Fig. 9 is far too low—much too dead—which we also probably surmised from the excessive padding added to the room.

Somewhere between the completely live room (128 units) and the over-subdued room (710 units), there must be a happy medium. Without going through the calculations, but following the same general method, we find that if we drape only one-fifth of the area of each wall and carpet the floor we arrive at a total absorption of 357 units (per the third calculation of Fig. 12). This room, now re-adjusted to 357 units, yields a reverberation time of .70 second. Although this figure is a little on the low side, it will insure a reasonable acoustic "liveness."

The room should actually be a bit more on the live side because if you put a couple of upholstered settees in the room (at an absorption figure of 5 units each) and sprinkle generously with seven or eight human beings (4 absorption units each), the total absorption will be gradually increased to approximately 400 units which will tend to pull down the reverberation properties of the room to near the lower "acceptable" limit.

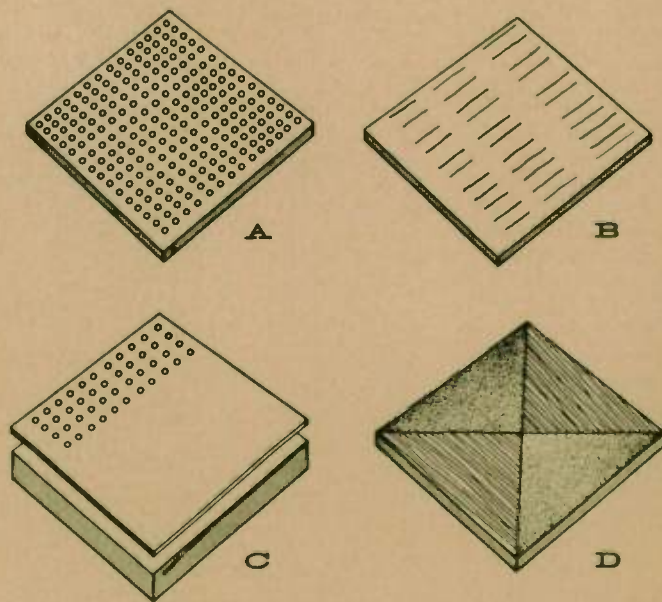


Fig. 13. Commercial acoustic tiles. (A) Soft fibers pressed into rigid, but acoustically soft mass with perforations to absorbent body; also with surface absorption. (B) Soft fibers pressed into rigid, but acoustically soft mass with surface cracked, or fissured, for added absorption. (C) Hard panel, such as Masonite, perforated and backed by thickness of absorbent material; no absorption by surface. (D) Same as (B) but with surface sculptured for decorative effect.

Altering the Room

As just demonstrated, much can be accomplished in the line of altering the reverberation of a room by judicious application of various household materials, especially those in the drapery category. Thus, for instance, if one were to hang draw drapes across the four walls of a room (of the dimensions of the one given in Fig. 12) and kept the floor linoleum-covered, as in a playroom, the room could measure from 2.5 seconds with the drapes completely retracted to .46 second with the drapes fully extended to cover the walls. With this type of arrangement, it would be fairly simple for the host to adjust his room to match the number of guests: For a lot of people—open the drapes and expose the walls; for an intimate soiree—draw the drapes. On the basis that each adult is equivalent to 4 absorption units, one could "calibrate" the drapery drawstring in units of number of guests.

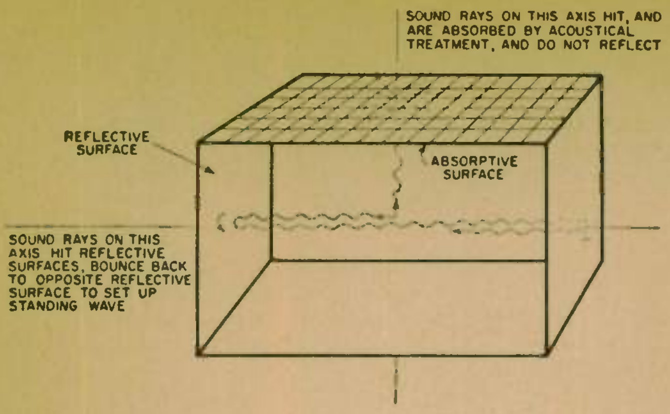


Fig. 14. Absorbent material on one surface will inhibit a resonance condition between that surface and the opposite one, but will not prevent resonance from being set up between the other room surfaces.

We may, of course, use materials which have been developed specifically for sound control, such as the acoustic tiles discussed previously. With these tiles we may control the room acoustics more accurately and with smaller areas of material than with conventional drapes, although combinations of tiles and drapes are practical. Tiles are especially useful when the room decor will suffer from an overabundance of drapery. Actually, the acoustic tiles may be quite decorative as well as functional and lend a clean, uncluttered look to the room. With the modern trend toward "airy" interiors, acoustic tile may actually prove a boon in maintaining this feeling while controlling room acoustics. Many such tiles (especially the perforated ones) can be painted to create interesting decorative effects.

One of the most important features of these tiles is that they enable us to treat ceilings as well as walls. Since the ceiling of a room comprises approximately one-sixth of the room's interior surface, it represents a good sized area which cannot be treated other than with tiles. The ceiling cited in our example, 25 feet x 20 feet with an absorption coefficient of .06 when bare has only 30 absorption units but when covered with fissured acoustic tiles its absorption becomes 375 units—more than enough to take care of the whole room without treating the other areas.

Undesirable Resonances

Although this method of adjusting room acoustics in one grand stroke is fine for general noise abatement, it is not the best approach to the problem of musical acoustics control. The reason for this is that there may be room resonances set up between the sets of untreated parallel walls which can produce quite disturbing effects to musical sounds reproduced within the room. Room resonances may occur when the dimensions of the room are close to the wavelength of the sound propagated in the room and are most prevalent in the low-frequency region.

When the air within the listening room is energized by a sound source, the volume of air enclosed by the walls, ceiling, and floor may be driven into resonance at frequencies which coincide with the wavelength or half wavelength separation of the walls. The very worst shape for a room, acoustically speaking, is a perfect cube because all axes are of equal length and the room is highly excitable acoustically and energetically resonant in all directions at the wavelength (and harmonics of it) which is equivalent to the spacing of the walls. In general, no room dimension should be close to a whole number ratio of another dimension. For normal living rooms, you may consider that if the dimensions of your room are in the ratio of 1:1.25:1.58 then your listening room will be reasonably immune to resonances between the three mutually perpendicular axes.

Although there is usually nothing we can do to the dimensions of our rooms, the statement of the condition, as given, is important in that it allows us to allocate our sound absorbent material intelligently. For instance, if there should be a strong resonant condition between two highly reflecting *walls*, then even large amounts of sound absorbing materials on the floor or ceiling will not affect that particular wall-to-wall resonance to any great extent. See Fig. 14. Thus the technique of treating the ceiling *only* is not the answer to adjusting the room for *music* purposes. We must distribute the "treatment" in random fashion to include all surfaces in order to minimize reflections along all three axes. If we thus distribute the treatment throughout the room, not only will we minimize room resonance but we will improve and equalize the general sound diffusion throughout the room—a condition of considerable importance for stereo.

Speaker & Listener Locations

One of the prime considerations in setting up a room for music reproduction—live or recorded—is that the

Table 1. Absorption coefficients per square foot of some common materials are shown.

ABSORPTION COEFFICIENTS

FLOORS	FREQUENCY (cps)					
	128	256	512	1024	2048	4096
Linoleum on concrete	.02	.02	.03	.04	.04	.04
Pine flooring	.08	.09	.09	.10	.10	.09
Carpet on ⅛" felt	.11	.14	.37	.43	.27	.25
Cork slabs, polished	.08	.08	.08	.19	.20	.21
WALLS AND CEILINGS						
Brick wall, unpainted	.02	.02	.03	.04	.05	.07
Brick wall, painted	.01	.01	.02	.02	.02	.02
Poured concrete, painted	.01	.01	.01	.02	.02	.02
Wood veneer on 2 x 3 studs 16" on centers	.11	11	.12	.11	.10	10
Plaster on hollow tiles	.01	.01	.02	.03	.04	.05
Plaster lime, sand finish on metal lath	.04	.05	.06	08	.04	.06
Pine wood sheathing	.10	.11	.10	.08	.08	.11
Window glass	.03	.03	.03	.02	.02	.02
WALL FURNISHINGS						
Cotton Fabrics 14 oz. yd. draped to ½ width	.07	.31	.49	.81	.66	.54
draped to ⅞ width	.03	.12	.15	.27	.37	.42
Velours 18 oz. yd. flat	.05	.12	.35	.45	.38	.36
draped to ½ own width	.14	.35	.55	.72	.70	.65
FURNITURE AND PEOPLE						
Upholstered mohair chair	2.5	3.5	4.5	4.5	4.8	5.0
All wood chair	1.8	2.0	2.4	2.4	3.8	2.5
Plastic covered chair	1.5	3.3	4.3	4.0	3.8	3.6
Child standing	1.8	2.2	2.8	3.8	3.5	3.5
Adult standing	2.5	3.5	4.2	4.6	5.0	5.0
Adult in upholstered chair	3.0	4.0	4.5	5.0	5.2	5.5
Woman in evening dress	1.8	2.0	2.3	3.6	3.6	3.5

sound source be located in a relatively "live" area so that the sound may be projected *into* the room. For monophonic applications, an undraped corner of the room is perhaps the best "projector" area of the room. Many articles have recommended locating the loudspeaker in a corner so that the corner can act as a three-sided horn *feeding* the room. Too little mention is made, however, concerning the acoustic condition of the walls and floors of these "desirable" corners. If the walls and floors are heavily draped and carpeted, as shown in Fig. 15A, the corner would be of very little use. The need for a live backing area behind the sound (Fig. 15B) is dictated by the need for throwing the sound forward—just as is done by an orchestral shell (Fig. 15C). Therefore, absorption techniques that are to be applied should be in the general listening area, with a minimum in the sound generating area.

Putting the absorbent material in the listening area serves to reduce unduly strident echoes from live surfaces (removed from the original source) which by reflectively interfering with the direct sound will cause confusion at the listener's ear. In this connection we must also realize that the closer one sits to the source of the sound, the less effective are the surroundings. This effect is especially noticeable in very live rooms. If one were seated close to a hard plaster wall and the sound source is 20 feet away, there would be considerable interference introduced by reflections off the wall. On the other hand, if the walls were heavily draped then the ear would receive very little reflected sound and its major acoustic stimulus would be direct sound from the speaker. We must, therefore, strive for a balance between direct and reflected sound so that while we hear the performance in front of us, we get some general diffuse sound from around us to "liven" the performance.

Summary of Principles

Summarizing the various principles outlined for any good listening room—whether for live music, reproduced monophonic or stereo, we have:

A. The room should have an absorption characteristic which will provide a reverberation time of from .75 to 1.25 seconds.

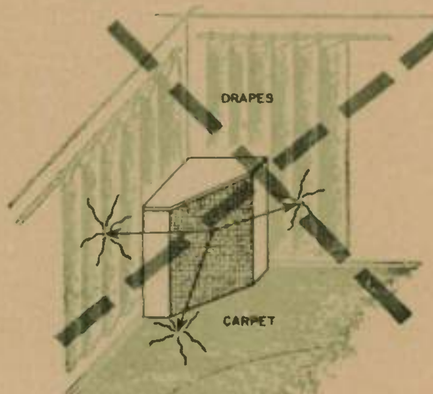
B. The sound absorption material used to achieve this reverberation figure should be distributed throughout the room rather than be concentrated in one area.

C. Large, hard parallel reflecting areas should be avoided as they lead to multiple reflections and resonances.

D. The sound source should be located in a comparatively "live" area of the room.

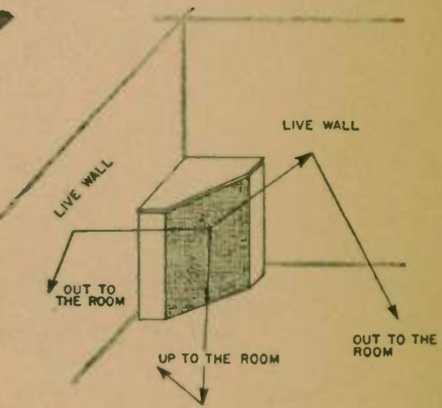
E. The listener location should not be adjacent to either a hard reflective surface or a totally absorptive surface since in neither case will the listener receive the proper proportion of reflected-to-direct sound in keeping with accepted reverberation practice.

Fig. 15. Speaker placement should be such as to provide proper "live" coupling.



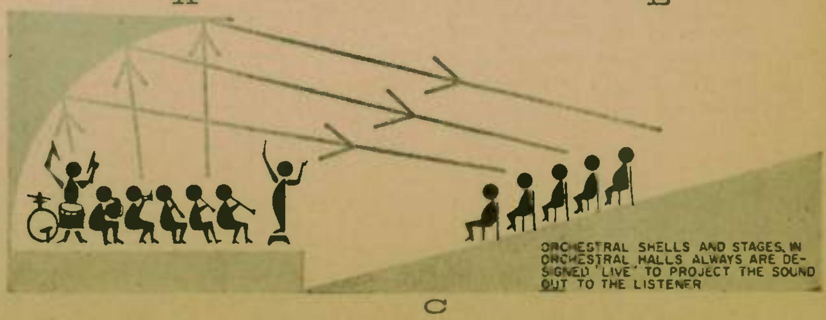
A CORNER IS USELESS FOR COUPLING THE SOUND TO THE ROOM IF THE CORNER SURFACES ARE SO PADDED THAT THEY ABSORB ALL THE SOUND THAT REACHES THEM

A



A CORNER WILL PROPERLY COUPLE THE SPEAKER TO THE ROOM IF IT IS LIVE ENOUGH TO PRODUCE SUFFICIENTLY STRONG REFLECTIONS SO THAT THE SURFACES ACT AS EXTENSIONS OF THE SOUND SOURCE ITSELF

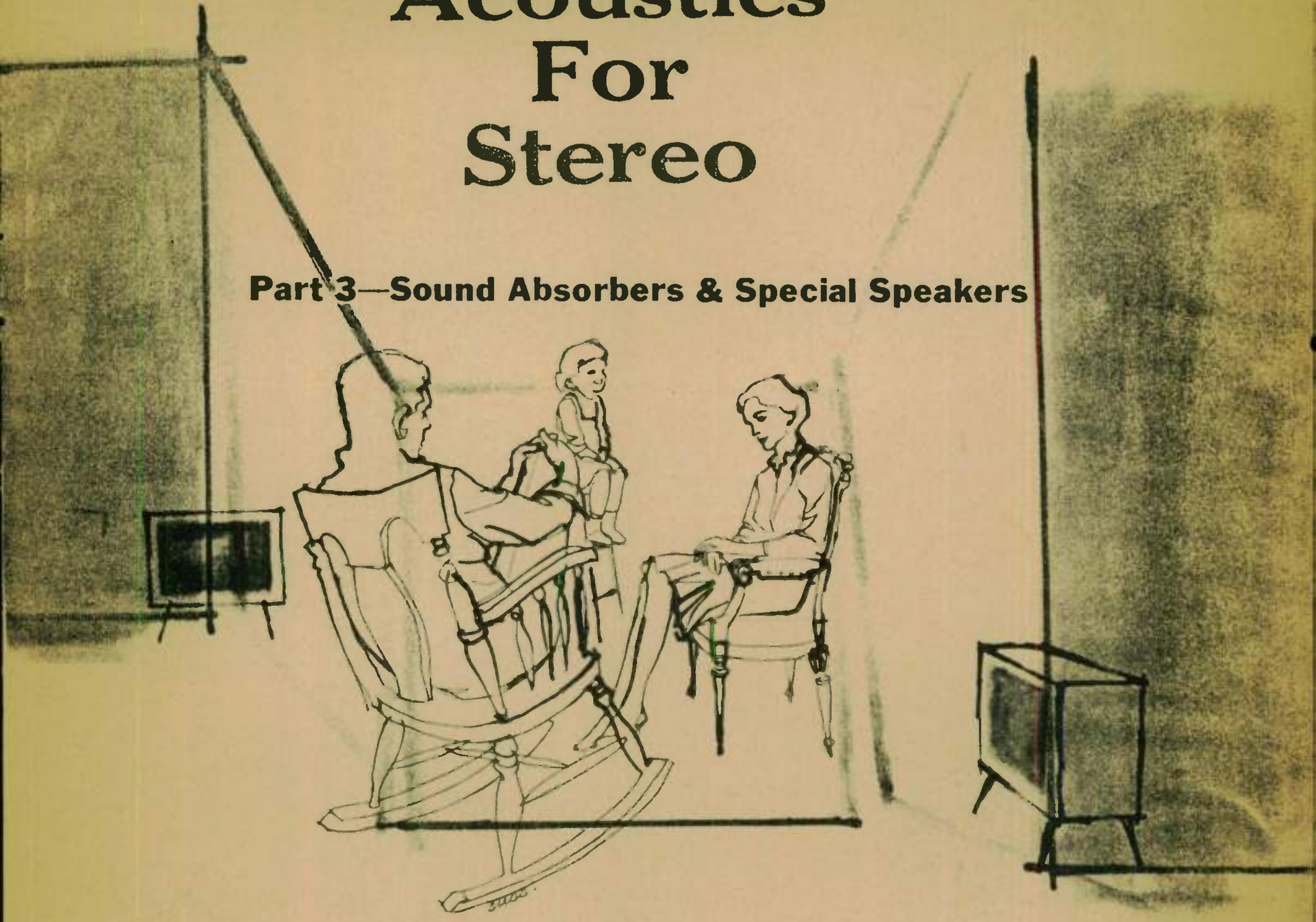
B



C

Room Acoustics For Stereo

Part 3—Sound Absorbers & Special Speakers



By **ABRAHAM B. COHEN** / Advanced Acoustics Corp.

Practical methods of tailoring room acoustics for the best stereo performance. Various speaker setups are covered.

THE static effectiveness of the acoustic condition of a room is determined by the room structure and its furnishings. However, it may be greatly affected by many factors such as the number of people in the room, the type of program being reproduced, and whether windows or doors leading into the room are open or closed. As discussed in Part 2 of this series, this condition can be kept under control by means of variable sound absorption devices which may be installed and adjusted at will to meet these variables as well as provide the proper acoustic environment for the various types of stereo speaker systems available today.

Room Acoustic Adjustment

In the earlier articles we discussed the possibility of providing variable control over room acoustics by means of retractable drapes, so that the overall reverberation time could be adjusted to take into account the number of people in the listening room at a given time.

If the reader wishes to make a simple test which will demonstrate the efficacy of such variations, without getting involved in physical changes in his room, let him listen to his sound system first with the windows open and then closed. If he has two average-sized windows in a room, say 3 feet by 6 feet each, this amounts to 36 square feet of glass with practically no absorption when closed. Next open both windows wide. If they are casement windows you can get a full 36 square feet of open space; if they are double-hung windows you will, of course, get only 18 square feet of open space. The open windows represent an additional 36 or 18 units of absorption. The acoustical effect of this added absorption should be readily apparent, especially if you place yourself not too far from the windows. From a practical standpoint much of our hi-fi will just "go out the window" of course.

Now, the amount of absorption produced by opening the room's windows is not much greater than that introduced by a small group of people. If you do nothing to compensate for their number, your audio demonstration will sound flat and dull, and you will try to push your system harder or boost the highs. The immediate reaction of the newcomer to hi-fi will be "Do you have

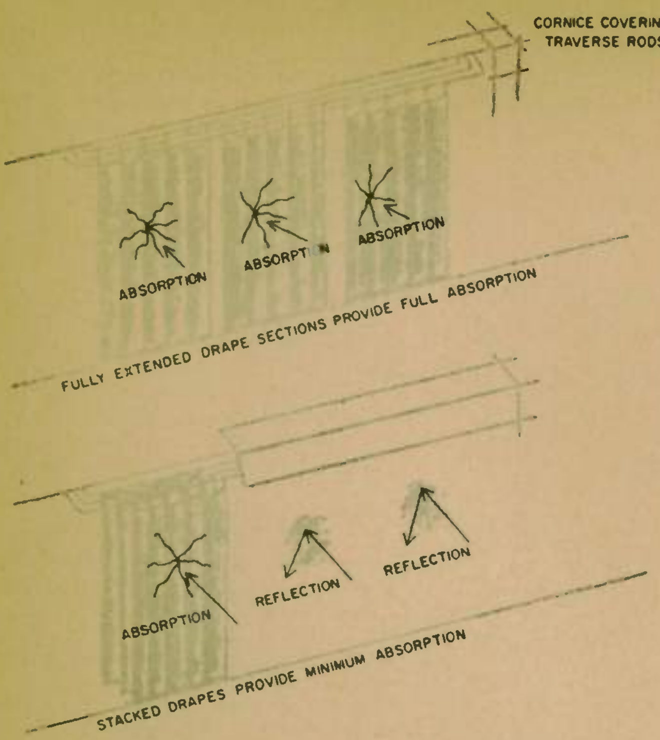


Fig. 16. Method of "stacking" drapes so that a wide variation of absorption control may be obtained to compensate for increased number of people in room, or to change the reverberant condition to suit the type of music that is being listened to.

to play it so loud?" The answer is "yes," in order to overcome the psycho-acoustical "smallness" of the sound due to the lack of proper reverberation which normally enlarges the sound. It is thus desirable to adjust the room on the live side with some means provided for its variation to accommodate changing interior room conditions.

In our earlier treatment of reverberation time as a function of room size it was indicated that the reverberation characteristics also varied for different types of musical reproduction. For example, the clean, curt articulation of fast musical passages with notes coming in rapid succession and literally falling all over one another, would be completely smeared into oblivion if they were to be reproduced in a room so reverberant that the sound of one note kept swirling around the room long after it was played and while a new note was trying to make itself heard.

Similarly, the articulation of a man's voice may be completely obliterated if he talks at a rapid fire rate in a highly reverberant auditorium. The ever-reflecting echoes would wash around the listener in such profusion that the original delivery might be completely obliterated. This same auditorium, however, might be admirably suited to slow, majestic organ music. The auditorium, with its long reverberation period, would allow ample time for the organ's characteristic slow phrases to be properly re-enforced. The pipes would "resound" through the spaciousness of the auditorium.

Because of variations in program material, no single room can be "optimum" for all types of sound reproduction. Sliding scales have been worked out by acoustic engineers correlating the general type of music to be heard with room size and optimum reverberation time. In many cases a good compromise can be reached in an architectural structure so as to accommodate a fairly wide range of musical performances with optimum results.

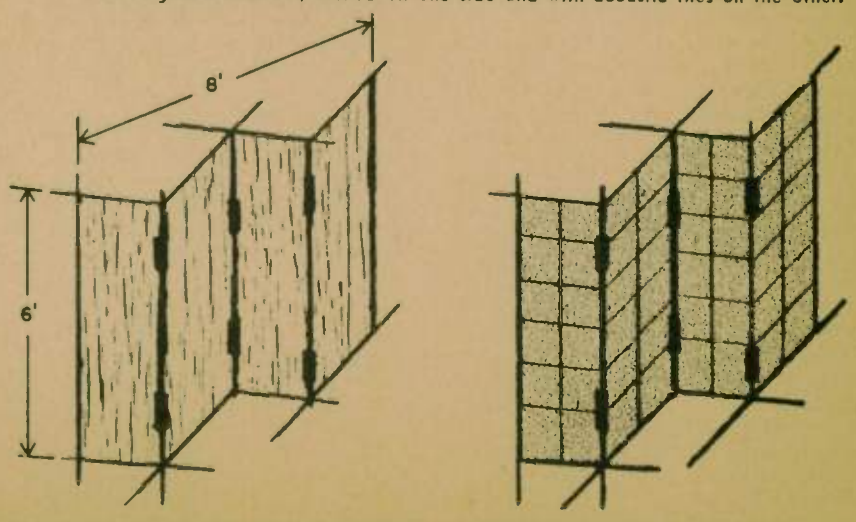
Where one strives for the ultimate in sound conditioning, then means for adjusting the room or auditorium to the type of performance are usually included in the structure. Specifically, recording or broadcasting studios frequently make use of such acoustically variable devices to insure optimum reproduction of a particular type of performance. Some of these professional techniques can be adapted for use in the home with gratifying results.

In order to have control over the acoustics of our listening room, it is best to start with a fairly "live" condition. Then, as more people come into the room the liveness will drop to the desired level or as the character of the music is varied we may alter the liveness of the room to suit the program by the addition of absorptive surfaces. One simple means of staying on the "live" side while providing the requisite flexibility is to use drapes mounted on traverse rods so that they may be pulled out straight or gathered in folds. With draperies, the home decorator does not find herself limited to specific areas which must be covered. She has a choice of materials and draping methods by which various reverberation combinations may be obtained.

Traverse-type, or drawstring, drapes are an excellent means of varying room acoustics to suit either the *type of music or the number of people* in the room. It is to be expected that friends and relatives will visit occasionally. If four people drop in we have added another 16 units of absorption to the room and even though the party may liven up socially, the room becomes deader acoustically. In fact, with four or five more people in the room, it would be desirable to remove the drapes entirely in order to maintain proper reverberation. Of course, removal is out of the question once they are up but there are means of manipulating them to get the proper reverberation correction as, for example, compressing them into a very small space—if they are hung on a traverse rod.

We may also adapt, in rather simple fashion, some of the proven techniques

Fig. 17. An approximate change in absorption of 6:1 may be obtained by use of a folding screen wood-panelled on one side and with acoustic tiles on the other.



WOOD PANELS ON ONE SIDE OF SCREEN
48(SQ FT.) X .12(ABS. COEF.) = 5.8 UNITS

ACOUSTIC TILES ON OTHER SIDE OF SCREEN
48(SQ. FT.) X .75(ABS. COEF.) = 36 UNITS

of the professional recording and broadcast studio. In broadcast studios reversible wall panels with different sound absorption characteristics are often used. Depending on the characteristic desired as a function of the size of the performing group and the placement of the individual sections of the group, these panels can be adjusted to provide the correct reverberation time. While we can't expect such an installation in the average home, in new houses being built for the serious audiophile there is no reason why some devices can't be incorporated. In existing structures we may employ an alternate method of varying the acoustic environment by "stacking" the drapes. This method, as illustrated in Fig. 16, relies on the number of layers in the stack to provide the variation. Thus, if we consider an area of 80 square feet of drapes (8 feet high by 10 feet wide) in the room previously discussed, broken up into three separate sections 3$\frac{1}{3}$ feet wide, each section capable of being pulled back of the others, then we have an option of the full 80 square feet or as little as 27 square feet of acoustical "treatment." For more or less flat-hung drapes, this would give us an absorption coefficient which could be varied from 80 x .15 = 12 units to one-third of that value or 4 units. This means that if the room is properly adjusted for your immediate family when the drapes are fully extended, should 3 or 4 more people (at 15 or 16 units of absorption) arrive, all you would have to do is retract the drapes and the acoustic environment is again "normal."

Another method of adjusting the room involves the use of folding room-divider-type screens. If these screens are wood panelled on one side and covered with an absorbent surface, such as acoustic tile, on the other, there will be available an approximate 6:1 ratio of acoustic absorption. As shown in Fig. 17, if the screen were 6 feet by 8 feet when opened, its wood-veneered side would provide 5.8 units of absorption (48 x .12) while the tiled side would yield 36 units (48 x .75), which for the small area involved is quite a respectable change. Still another change can be obtained by applying acoustic tiles to doors which lead into the listening room. If such doors are of standard size (2$\frac{1}{2}$ feet by 7 feet), when covered with acoustic tile they provide 13 absorption units (17.5 x .75). Such panels may be made reversible by installing them in recessed grooves (see Fig. 18A) so that the panel may be slipped in with the tiled side outward or the veneered side showing, as dictated by existing conditions. Or, such panels may simply be hung from hooks from the top of the door frame, as shown in Fig. 18B, and are thus quickly reversible. Where metal clad doors are involved, simple strap hooks that go over the top of the door may be used. See Fig. 18C.

Finally, for the handy householder there is the semi-professional installation consisting of pivoted, double-faced

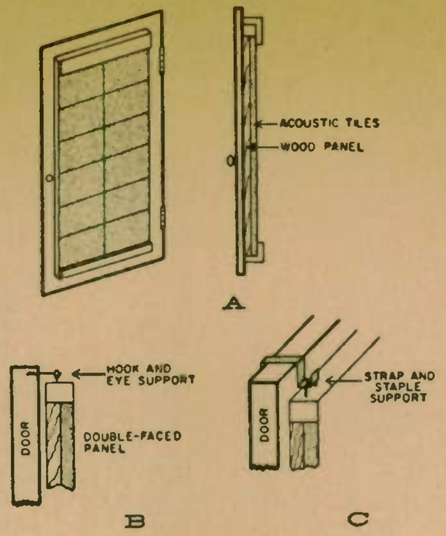


Fig. 18. A double-sided reversible panel applied to door area may provide either 13 absorption units when the tile side faces out, or 1.5 absorption units when the wood panelled side faces out. Some supporting methods are indicated in figure.

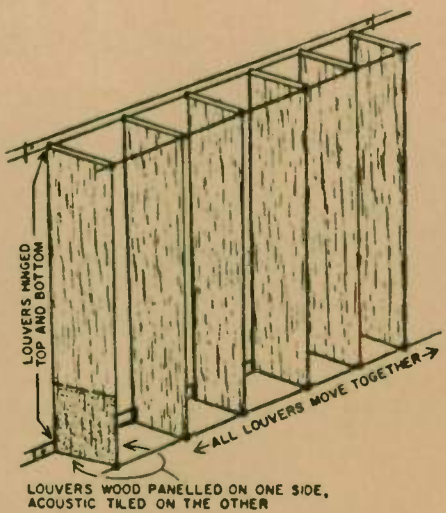


Fig. 19. A reversible louver assembly can give a wide variation of acoustical control over the reverberation of a room in proportion to the size and orientation of the louvers that are used. See text.

louvers (as shown in Fig. 19) with acoustic tiling on one side and wood veneer on the other. These louvers could be assembled on a framing panel and hung as a unit. The absorption by this method is dependent, of course, on the size and number of the louvers and their positioning. A second effect, and an important one, is obtained with these pivoted louvers. By leaving them in a semi-open position, rather than in a totally closed condition (in either direction), a more random diffusion of sound is obtained, thus minimizing standing waves within the room.

While we have spent considerable space on how to correct the over-all acoustic condition of the room by the addition of effective absorbent materials, it should be remembered that to be effective these treated sections should be scattered in a random fashion throughout the room so that one end of the room isn't completely live and the other end completely dead. Thus, in the matter of drapes, rather than have one solid area 10 or 12 feet wide, it would be better to divide the drapes into smaller sections of, perhaps, 3 to 4 feet wide and place them on more than one wall.

This same principle should hold when applying the tiles. Not only will this random application of the treatment break up inherent and troublesome room resonances, but it will permit us to balance the room for stereo program material and for the different types of speaker systems which might be adopted.

Balanced Speaker System

In acoustically balanced rooms, that is, where the acoustic effect from the listener's left is the same as that from his right, due either to the original room condition or to its adjustment, we will discover that more than one type of speaker system will perform satisfactorily. The speaker systems, shown in Figs. 20A, 20B, and 20D (Fig. 4, Part 1, Jan. issue and reproduced in this part) are all essentially *balanced* systems although they differ from one another in design concept.

In Fig. 20A, although there is no center speaker, an apparent "middle" is heard when speaker phasings are correct and when the program content of each channel includes judicious amounts of program from the other. The overlapping *common* program content from one channel to the other produces an apparent sound field located between the two speakers. In order to obtain optimum "center fill" to heighten the stereo effect of this balanced-speaker system, the speakers must be backed by a live, hard surface which will then function as a re-radiating device. Moreover, in a system such as this balanced-speaker layout, the "center fill" becomes more apparent when the speakers are angled in towards the listener rather than flat against the wall.

This effect occurs because by angling the speakers, more of the high frequencies of each speaker field overlap at the listener's post. When these overlapping higher-frequency fields contain common program material at equal level, their apparent source will be directly in front of the listener. Obviously, then, the speaker separation and the common angle between them will not only determine the maximum area of equal stereo perception but will also be indicative of the degree of apparent center fill.

Phantom Channel System

In the case of the system diagrammed in Fig. 20B, where there is an actual center speaker delivering a third signal from both channels and consisting of common program com-

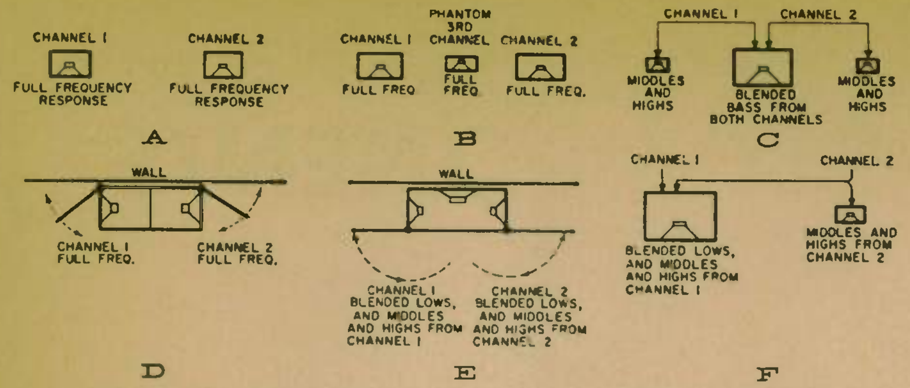


Fig. 20. The wide variety of stereo speaker systems in use. (Repeated from Part 1.)

ponents in both, speaker separation and corresponding angles are not nearly as critical as in the previous case. The "center fill" is always present irrespective of positioning or angling of the main channel speakers. Obviously, the system employing a phantom channel may be spread out over a greater wall area than would be the case with a simple two-speaker system. However, as the system is stretched out, the full-range speakers will have to be toed in centrally otherwise the stereo-determining high-frequency beams from the end speakers will not overlap sufficiently to produce a large enough stereo listening area. In an effort to widen the wall of sound by stretching out the phantom system, it is necessary to insure that the wall back of such a system is acoustically uniform. It would not do to have areas of absorbent material on the wall behind such speaker cabinets. Such absorbent areas would, by their acoustic characteristics, create soft spots that would not reflect the sound sufficiently to guarantee the continuity of the sound wall from one source to the other.

This latter condition can be achieved by putting the end speakers in the corners of the room. In Part 2 of this series, we showed that for maximum sound radiation from a corner, the corner should be live. If the wall adjoining the corner enclosure is acoustically soft, there will be no apparent sound source reflected from the wall and radiation will be reduced.

Carrying this idea to its conclusion, as shown in Fig. 21, we see that soft spots intervening between the speakers will create discontinuity in the wall of sound while hard reflecting surfaces will help merge the sound sources into a blended wall of sound. We again arrive at the same conclusion regarding room acoustics for this system as we did for the simple, two-speaker system. Even though capable of greater extension and greater uniformity of sound, we still need a completely live backdrop, or shell, from which the system's sound may be projected into the room.

Back-Hinged Door System

The system that is possibly least dependent on environmental acoustics is that of Fig. 20D for when the side doors (hinged at the back) are angled to throw the sound into the center of the room, the speakers see the individual hard reflecting surfaces of the doors as live reflecting areas. Even when opened up wide against the walls, these doors will act as adjacent hard reflective surfaces for the sound that hits them, thus adding some degree of liveness to the reproduction.

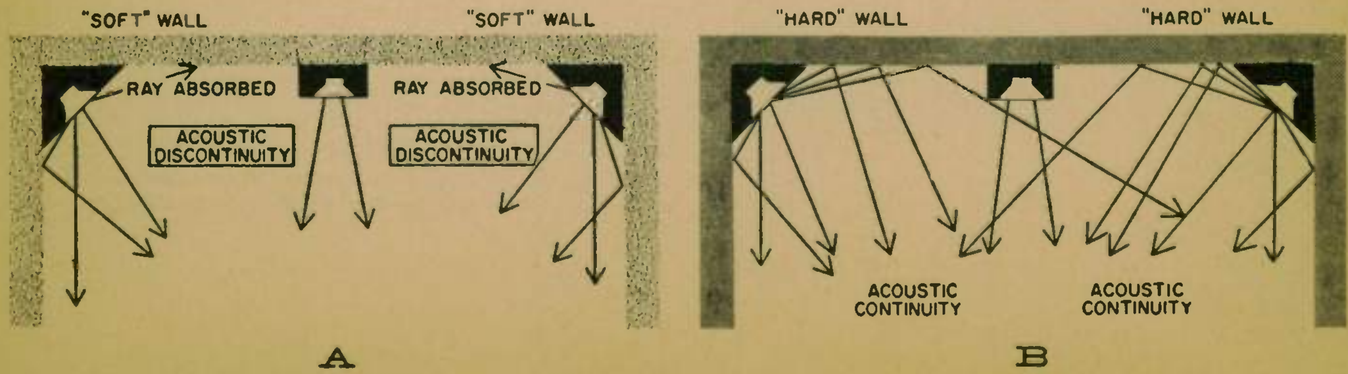
Although for general reverberation conditions it may be desirable to keep the over-all back wall as live as possible, some stereo effect may be lost with this hinged-door system. When only the doors serve as the reflecting surface, then the direction of the reflection and the subsequent stereo effect becomes a function of the door angle. Where, however, a completely live wall backs up the doors of the cabinet, then there is reflection from both the door *and* the wall and control of the stereo effect by door manipulation is reduced.

Satellite System

The system of Fig. 20C, where the lows from both channels are combined in one speaker which, in turn, is flanked by higher frequency satellite speakers, perhaps lends itself to greater decorative flexibility than the other systems just described. For the balanced bass woofer enclosure we want a live wall behind the system. However, the fact that the satellite speakers carry only the high-frequency program components indicates that their sound fields will be more "beamed" than will the blended bass from the common woofer, especially if the satellites are horn-type speakers. Where such beaming occurs, there will be a minimum of rear sound wave projection which means that no special acoustical treatment need be given the wall areas behind the satellites since there will be no sound field to be reflected.

This preponderance of directly beamed sound from the satellites means that the walls behind the satellites can be decorated in any manner, leaving only the area surrounding the common woofer live enough to bounce the sound out into the room. The satellites may be removed some distance from the common woofer before a discontinuity of sound will be noticed between the satellite and common woofer. Psycho-acoustically, we take the fundamental lows from the common woofer and add them to the higher frequency components from the satellites to produce the illusion of the entire program coming from the satellites. However, if we stretch the system too much and/or get too close to the satellites, there will be a discontinuity of channel content. As in previous examples, when stretching out such a system in an attempt to enlarge the wall of sound, then it will be helpful to increase the live wall area

Fig. 21. (A) When speaker systems are spaced far apart with absorbent areas between them, there is little sound reflected from these absorbent areas and there results a discontinuity in the "wall of sound." (B) A reflective wall produces apparent sound sources between the spaced systems and improves the continuity of the "wall of sound" between the loudspeaker systems.



to cover not only the common woofer but to reach out to the satellites as well. This live wall, by creating *new* apparent low-frequency sound sources, will help to "bridge" the lows from the blended source over to the satellites and smooth out the wall of sound. It is also recognized that as the satellites are moved farther from the center they should be angled in more and more to provide maximum overlapping areas of the high frequencies.

Front-Hinged Door System

Perhaps the system most dependent on room acoustics for its stereo performance is the one shown in Fig. 20E. Due to the fact that the common woofer faces the rear, unless this wall is quite live the lows will be overly subdued. Certainly if there is any draping of this area the lows will be unduly absorbed. For this system, then, a hard, highly reflective back wall is definitely prescribed for maximum low-frequency performance. However, we are more directly concerned with the orientation of the high frequencies which emerge from the sides of the cabinet and reach the listener *via* both wall-reflected paths and diffraction paths around the doors. Because of the direction and extended reflections of such paths, the stereo effect is obtained. For optimum effect from such a system, not only will the entire back wall have to be live but a good part of the side walls of the room must be live in order to produce the reflected rays that will convey the stereo effect to the listener.

Single Satellite System

The system of Fig. 20F can now be analyzed in terms of the systems just discussed. In this system, the lows are again blended in one system and the highs from one channel also radiate from that same system. The highs from the second channel, however, emerge from a single satellite speaker. This system may be considered as one-half of the system shown in Fig. 20C. The lows from the common woofer are psycho-acoustically bridged over to the highs of the satellite and the entire second channel thus appears to come from the satellite. The satellite may be considerably removed from the main enclosure before any disembodiment of the program content occurs. However, such disembodiment can be minimized by keeping the wall behind the main enclosure and the satellite live so that new low-frequency sound sources induced in the live wall between the woofer and the satellite may help bridge the lows to the satellite even though the satellite is considerably removed (within reason) from the main enclosure.

Thus we have analyzed various types of speaker systems and their effect on the listener and the effect of the acoustical condition of the room upon such speaker systems. The final article in this series will cover some actual room arrangements for optimum stereo reception.

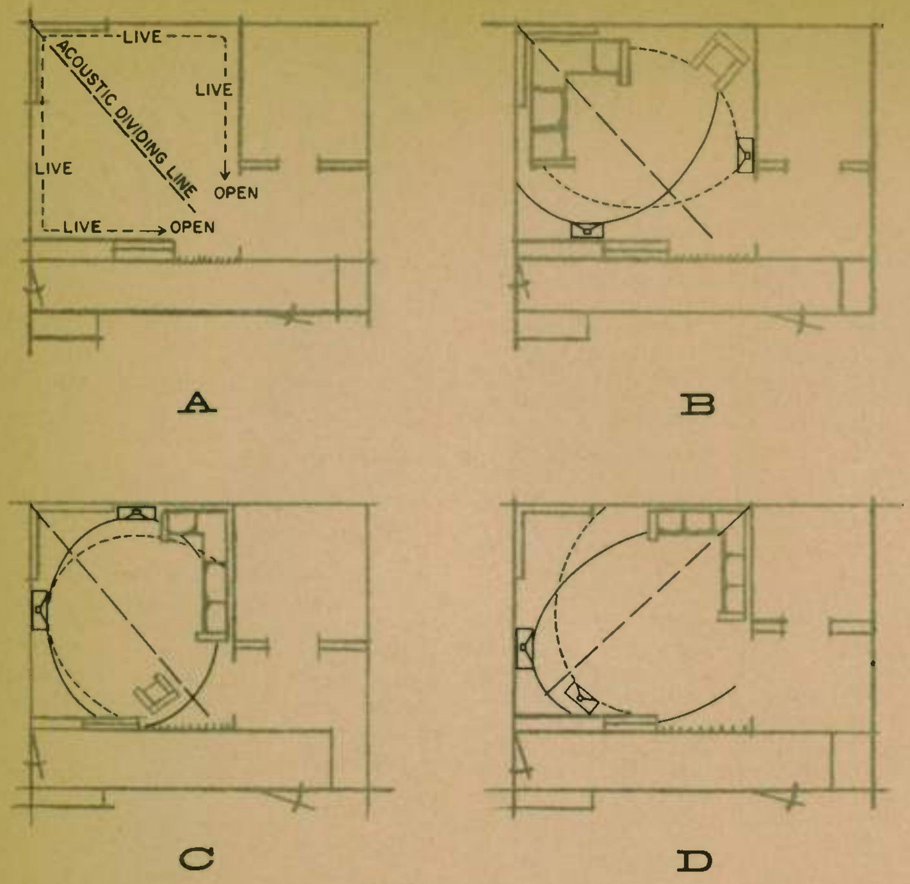


Fig. 24. Various means of arranging the listening room for stereo reproduction.

that there is seating capacity for five persons. This will be the equivalent of five mohair-upholstered chairs which individually have an absorption characteristic of 4 units, making a total of 20 units which will be added. Most likely there will be a rug on the floor and we will assume wall-to-wall carpeting in the living room only, leaving the bare parquet floor in the dining area. The 512-cps absorption coefficient of carpeting on a felt pad is approximately .37. The *total absorption* for the rug covering the 180 square feet of the living room floor is 180 x .37 = 66 units. Adding these 66 units of carpeted floor area to the 20 absorption units contributed by the chairs, brings the newly added absorption up to 86 units. We now have approximately 110 − 86 = 24 units of additional absorption to be discretely applied throughout the room to provide acceptable reverberation and stereo acoustic balance. To be perfectly accurate, we have more than that amount to play with. For instance, suppose we do hang some drapes. By actually hanging such drapes we mask off some of the wall area and thus obtain a sort of see-saw balance between new absorption material being hung as against the original material being covered up. However, where the original bare area has a very low absorption characteristic, such as plaster with a factor of .06, which is then covered with drapes with a coefficient of .5, then the absorption of the original wall may be neglected when masked by the drapes.

We now have 24 units to be deployed around the room. Some of these 24 units will, of course, be absorbed by the people in the room. The average absorption (acoustic) of an adult is approximately 4 units and for a child 2.5 units. If our "stereo evenings" are comparatively small family affairs, we may assume two adults and two children in the room for a total absorption of 13 units. This leaves approximately 11 units to be "installed."

The lady of the house will have no doubt noted the absence of curtains and/or drapes. As far as light gauzy curtains are concerned, these may be used in any quantity desired for they contribute practically nothing in the way of acoustic absorption. Thus, even if the entire window area of the room is hung with sheer curtains, the absorption of that area would still be governed by the absorption characteristic of the glass.

When it comes to the heavier materials, such as velours, monks cloth, or weighted cotton fabrics, they do, of course, contribute to the room's absorption. For instance, if we were to hang cotton drapes over a total of 10 linear feet (with the drapes 8 feet long) then we would have 8 x 10 = 80 square feet of drapes whose mid-frequency (512 cps) absorption coefficient is .15 when stretched out to ⅞th of their width. Then the total absorption of the drapes would be 80 x .15 = 12.0 units. Now we see that we have arrived at about the correct room adjustment.

Symmetrical Listening Axis

If the room is completely symmetrical (as is the case in Fig. 22), the problem of stereo balancing is fairly simple. We draw an imaginary line down the center of the room and proportion our absorption material in equal quantities on either side of the room (but still in random fashion). This will produce a situation where, when the stereo speakers are symmetrically placed with respect to the center balance line, they will "see" essentially the same acoustic environment on either side, with the result that the sound reflections from one wall will be practically the same as from the other wall; the spatial distribution of the sound within the room will thus be "stereo balanced."

Where, however, the room is not symmetrical, as in the problem we are treating, we must draw our imaginary line through the contemplated center of the listening room area, then place the speakers on either side of this line and balance both sides acoustically by adding or removing absorbing areas from either side to bring about an acoustic stereo balance.

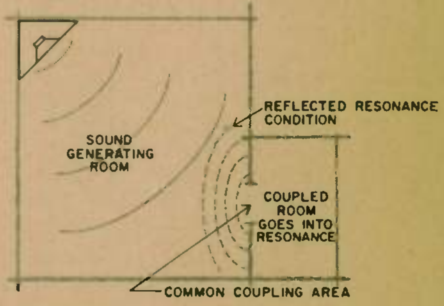


Fig. 25. Sound entering the door of the coupled room may cause the room to resonate and to inject this resonance back into the room in which the listening is done.

Referring to the diagram of our irregular room in Fig. 23, we can see that if we can find a line which bisects the room on an acoustically symmetrical basis, the stereo system can be placed accordingly. Then, as an extreme example, we can select another direction within the room which leaves the room unbalanced acoustically—and try to bring it into balance.

Referring to Fig. 24A, we have drawn such an acoustic dividing line from the corner casement window toward the dining area (there are other dividing lines which will be discussed in turn). It will be noted that there is fairly good geometric and acoustic symmetry about this line. There are two triangles of fairly equal acoustic influence on either side of this listening line. They have similar live walls opposite each other and then both end in an open area at the dining "L". We must now to cover not only the common woofer but to reach out to the satellites as well. This live wall, by creating *new* apparent low-frequency sound sources, will help to "bridge" the lows from the blended source over to the satellites and smooth out the wall of sound. It is also recognized that as the satellites are moved farther from the center they should be angled in more and more to provide maximum overlapping areas of the high frequencies.

Front-Hinged Door System

Perhaps the system most dependent on room acoustics for its stereo performance is the one shown in Fig. 20E. Due to the fact that the common woofer faces the rear, unless this wall is quite live the lows will be overly subdued. Certainly if there is any draping of this area the lows will be unduly absorbed. For this system, then, a hard, highly reflective back wall is definitely prescribed for maximum low-frequency performance. However, we are more directly concerned with the orientation of the high frequencies which emerge from the sides of the cabinet and reach the listener *via* both wall-reflected paths and diffraction paths around the doors. Because of the direction and extended reflections of such paths, the stereo effect is obtained. For optimum effect from such a system, not only will the entire back wall have to be live but a good part of the side walls of the room must be live in order to produce the reflected rays that will convey the stereo effect to the listener.

Single Satellite System

The system of Fig. 20F can now be analyzed in terms of the systems just discussed. In this system, the lows are again blended in one system and the highs from one channel also radiate from that same system. The highs from the second channel, however, emerge from a single satellite speaker. This system may be considered as one-half of the system shown in Fig. 20C. The lows from the common woofer are psycho-acoustically bridged over to the highs of the satellite and the entire second channel thus appears to come from the satellite. The satellite may be considerably removed from the main enclosure before any disembodiment of the program content occurs. However, such disembodiment can be minimized by keeping the wall behind the main enclosure and the satellite live so that new low-frequency sound sources induced in the live wall between the woofer and the satellite may help bridge the lows to the satellite even though the satellite is considerably removed (within reason) from the main enclosure.

Thus we have analyzed various types of speaker systems and their effect on the listener and the effect of the acoustical condition of the room upon such speaker systems. The final article in this series will cover some actual room arrangements for optimum stereo reception.

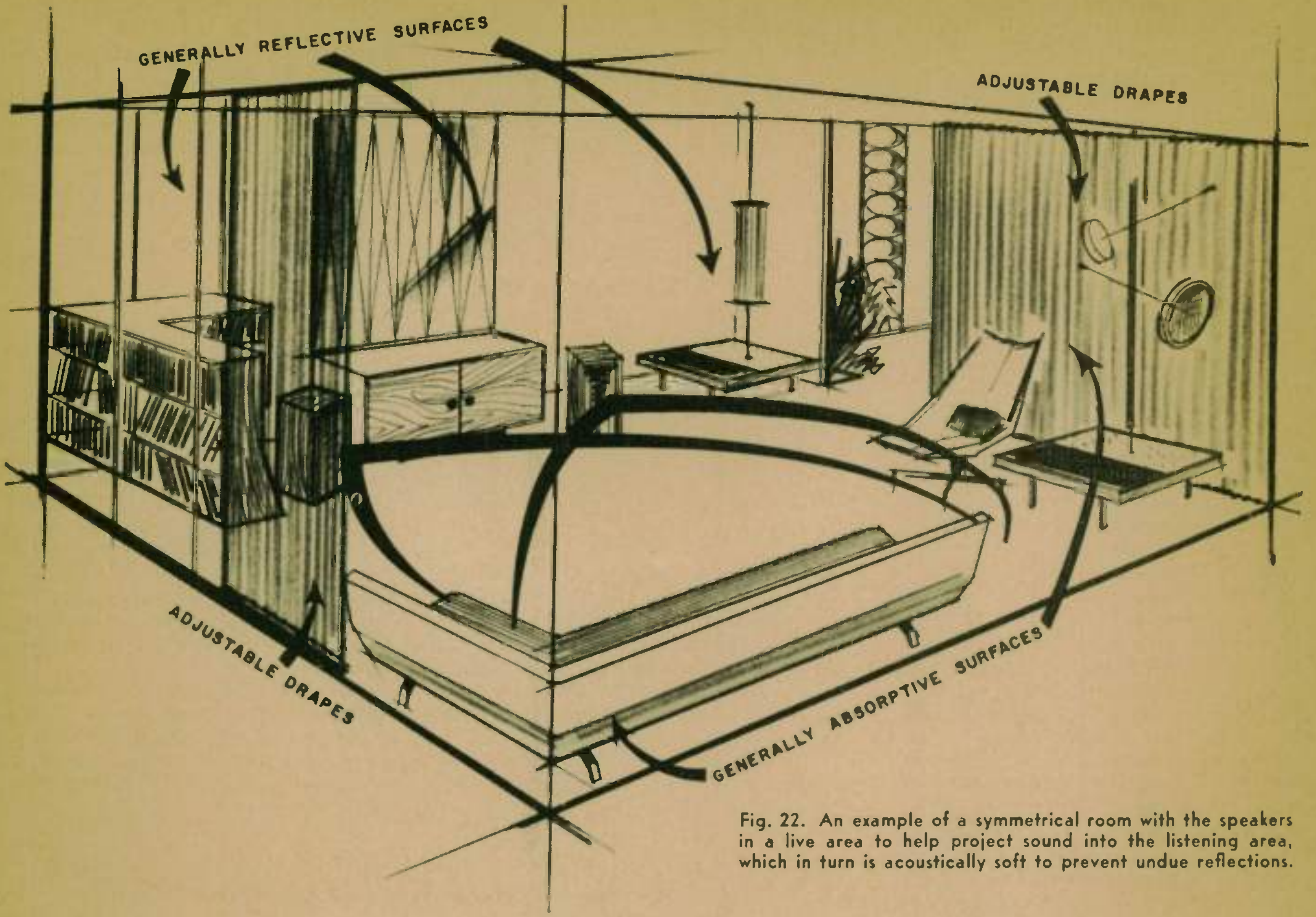


Fig. 22. An example of a symmetrical room with the speakers in a live area to help project sound into the listening area, which in turn is acoustically soft to prevent undue reflections.

Room Acoustics for Stereo

Part 4. Balancing the Room

By **ABRAHAM B. COHEN** / Advanced Acoustics Corp.

Practical techniques that can be used for proper room adjustment in order to get the best results from your stereo speaker setup.

IN THE first three articles of this series we discussed the basic principles involved in setting up a good acoustic environment for music: room treatment and reverberation; and variable sound absorption devices and special speaker setups. Now we will take up the matter of balancing the room.

A room which is physically balanced may need only general reverberation control while a physically unbalanced room may require a decidedly different wall treatment to provide proper balance for stereo listening purposes. The final treatment of the room and the placement of the furniture should be based on the establishment of a symmetrical listening axis from the speakers to the listener so that the same apparent acoustic characteristic will be presented on either side of the listener. Adjacent rooms leading into the main listening room, if not acoustically subdued internally, may act as coupled resonators, reflecting troublesome low-frequency resonances back into the listening room.

Physically Balanced Room

Irrespective of actual room configuration and the treatment throughout the room or the speaker system used, the end result should be such that one side of the room will produce the same acoustic reaction on the listener's ears as the other side of the room. Two interdependent conditions will exist. In one instance, the reaction of the room's acoustics *upon the listener* will determine what he hears while in the second instance the reaction of the room *upon the speaker* will determine how well it radiates or projects its sound.

In rooms which are physically symmetrical, the room is fairly simple to adjust to stereo requirements. In such a case we simply make sure that the room reverberation falls within the acceptable limits as outlined in the earlier parts of this series: *i.e.*, whatever absorbent treatment is applied is randomly distributed throughout the room and that the speakers are located at one end or wall of the room which is somewhat more "live" than the rest of the room. This simple situation, using balanced speaker systems, is illustrated in Fig. 22. We can calculate the total absorption by multiplying each discrete area by its own specific absorption coefficient, not forgetting, of course, the lumped total absorption of the chair, divan, and people in the room.

If these figures work out right for the cubic volume of the room, then we will have the right reverberation time. We will, in this instance, try to keep a reasonable minimum of absorbent material at the location of the speakers since we want a somewhat "live" proscenium from which the sound can bounce out into the listening area.

In this latter area—the listening area—we want the optimum (not maximum) sound absorption so that we will hear the correct proportion of reflected-to-direct sound to produce the correct liveness in the reproduction. In the present instance, placing the speakers against the hard wall with the highly reflective wall "mirror" behind them may add just the right amount of reverberation to the loudspeaker "proscenium" area.

The lounge chair, bookcases, distributed drapery, and floor covering may provide sufficient randomly distributed absorption area to properly diffuse the sound so that no acoustic "hot spots" will occur. In a room such as this, we may put our balanced speakers the conventional six to eight feet apart and toe them in, about 15 degrees toward the axis. The room acoustics will be fairly evenly balanced on either side of the listening axis and the room will be quite good for stereo listening, both ears being subjected to the same "liveness" from both sections of the room and full program content of each channel diffusing into the balanced room.

The Unbalanced Room

Now let's examine the situation with regard to the more irregular room layout and then apply our basic rules to obtain an acceptable stereo environment. It must be recognized that complete solutions for *all* types of systems would involve a complex analysis covering innumerable situations which is clearly beyond the scope of this series. Each system must, in effect, be "played by ear," keeping in mind the basic rules of good acoustics. If these rules are followed, then the listener should encounter a minimum of trouble in balancing his system, whatever it may be, with the room.

In order to demonstrate the technique involved, we will revert to our two balanced and similar speaker systems and try to optimize their stereo placement in the irregular room of Fig. 23. This is a type of room layout typical of many large apartments in many new developments. As one walks into the apartment through the door there is, to the right, a large area of plastered wall extending toward the far end where there is a closet and a door leading to another room. On the left there is a short stretch of plastered wall, then an open area leading to the living room which is dropped two steps below the hall level.

Farther along this direction, the area is still open but there is a railing divider between the central hallway and the dining alcove on the left. Stepping down into the living room, on the left there are built-in bookcases to the ceiling and then a bare plaster wall. In the corner is a rather large casement window, then two more plaster walls terminating in the dining alcove with a window at its wall side and the usual opening to the kitchen area through two room-divider elements between the dining "L" and the kitchen. The ceiling is 8 feet high, the central hallway floor is linoleum covered, and the other floors are wood parquet.

Our problem is to first determine the original bare-room acoustic condition then see how the room can be corrected to improve its over-all absorption, then distribute the absorption so that it makes a maximum contribution to the stereo acoustical balance of the room, and finally, to examine the various furniture arrangements to provide a suitable stereo listening environment.

Adjustment for Reverberation

From the dimensions of the room, its bare walls, ceiling, flooring, panelling, and windows and with the help of Table 1 (Part 2), which lists the absorption coefficients for these various building surfaces, it is not too difficult to determine the total absorption of the room.

Without going through the calculations, it turns out that the room has an absorption characteristic of 40 units. Entering this figure in the equation for the room reverberation constant, we get $T = .05V/A = 3$ seconds, where the total room volume is nearly 2400 cubic feet and $A = 40$.

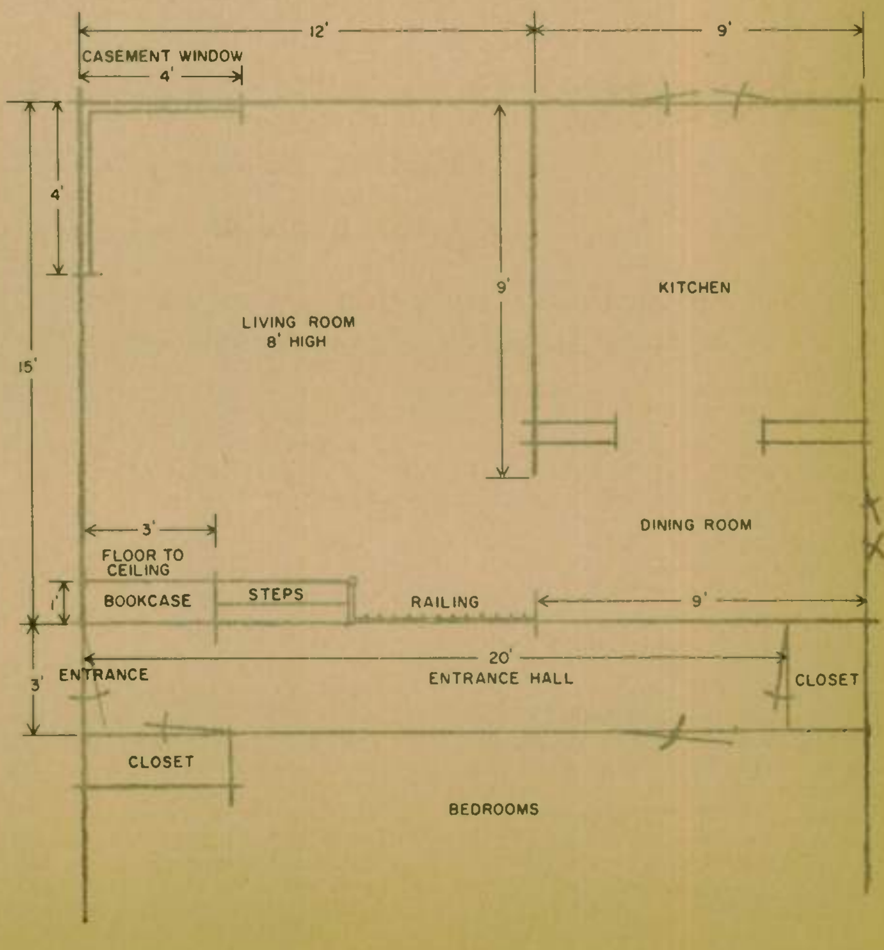
From our intuitive knowledge of completely bare enclosures, we know that the room will be too live, a hunch that is borne out by the 3-second figure which is far too high. Our problem now is to bring the reverberation down to "size" and to balance the room acoustically.

In order to strike a happy medium, let's try for an absorption of, say, 150 units which for this size room will bring the reverberation time down to .8 second: $T = (.05 \times 2400)/150 = .8$ second.

Incidentally, the kitchen area was not included in arriving at either the "room volume" or the absorption. The reason is that we have treated the kitchen area as a "coupled acoustic area" rather than as an integral part of the area being controlled. We shall be more specific about this matter of "coupled areas" after we have gone through the necessary acoustic corrections for the room.

To correct the room we will need 110 additional absorption units (150-40). Some of this absorption will naturally be contributed by sofas, settees, upholstered chairs, etc. We will assume

Fig. 23. Layout of the listening area and surrounding rooms discussed in the text.



of the drapes would be $80 \times .15 = 12.0$ units. Now we see that we have arrived at about the correct room adjustment.

Symmetrical Listening Axis

If the room is completely symmetrical (as is the case in Fig. 22), the problem of stereo balancing is fairly simple. We draw an imaginary line down the center of the room and proportion our absorption material in equal quantities on either side of the room (but still in random fashion). This will produce a situation where, when the stereo speakers are symmetrically placed with respect to the center balance line, they will "see" essentially the same acoustic environment on either side, with the result that the sound reflections from one wall will be practically the same as from the other wall; the spatial distribution of the sound within the room will thus be "stereo balanced."

Where, however, the room is not symmetrical, as in the problem we are treating, we must draw our imaginary line through the contemplated center of the listening room area, then place the speakers on either side of this line

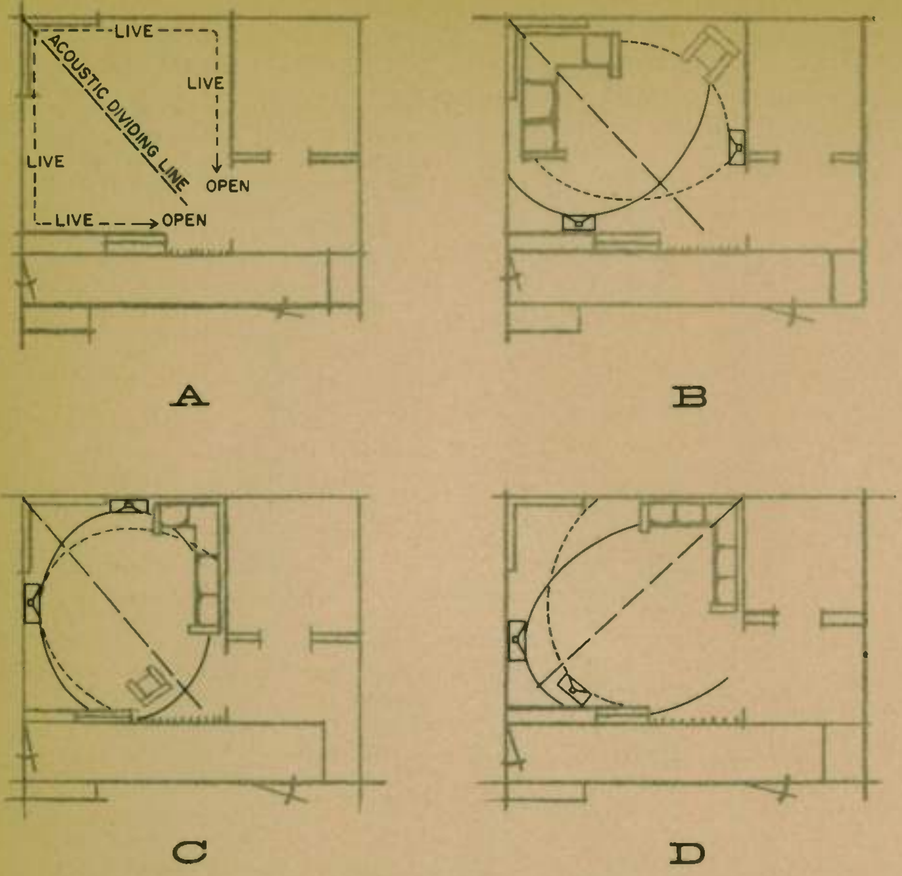


Fig. 24. Various means of arranging the listening room for stereo reproduction.

that there is seating capacity for five persons. This will be the equivalent of five mohair-upholstered chairs which individually have an absorption characteristic of 4 units, making a total of 20 units which will be added. Most likely there will be a rug on the floor and we will assume wall-to-wall carpeting in the living room only, leaving the bare parquet floor in the dining area. The 512-cps absorption coefficient of carpeting on a felt pad is approximately .37. The *total absorption* for the rug covering the 180 square feet of the living room floor is $180 \times .37 = 66$ units. Adding these 66 units of carpeted floor area to the 20 absorption units contributed by the chairs, brings the newly added absorption up to 86 units. We now have approximately $110 - 86 = 24$ units of additional absorption to be discretely applied throughout the room to provide acceptable reverberation and stereo acoustic balance. To be perfectly accurate, we have more than that amount to play with. For instance, suppose we do hang some drapes. By actually hanging such drapes we mask off some of the wall area and thus obtain a sort of see-saw balance between new absorption material being hung as against the original material being covered up. However, where the original bare area has a very low absorption characteristic, such as plaster with a factor of .06, which is then covered with drapes with a coefficient of .5, then the absorption of the original wall may be neglected when masked by the drapes.

We now have 24 units to be deployed around the room. Some of these 24 units will, of course, be absorbed by the people in the room. The average absorption (acoustic) of an adult is approximately 4 units and for a child 2.5 units. If our "stereo evenings" are comparatively small family affairs, we may assume two adults and two children in the room for a total absorption of 13 units. This leaves approximately 11 units to be "installed."

The lady of the house will have no doubt noted the absence of curtains and/or drapes. As far as light gauzy curtains are concerned, these may be used in any quantity desired for they contribute practically nothing in the way of acoustic absorption. Thus, even if the entire window area of the room is hung with sheer curtains, the absorption of that area would still be governed by the absorption characteristic of the glass.

When it comes to the heavier materials, such as velours, monks cloth, or weighted cotton fabrics, they do, of course, contribute to the room's absorption. For instance, if we were to hang cotton drapes over a total of 10 linear feet (with the drapes 8 feet long) then we would have $8 \times 10 = 80$ square feet of drapes whose mid-frequency (512 cps) absorption coefficient is .15 when stretched out to ⅞th of their width. Then the total absorption

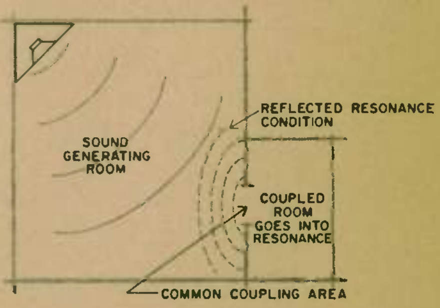


Fig. 25. Sound entering the door of the coupled room may cause the room to resonate and to inject this resonance back into the room in which the listening is done.

and balance both sides acoustically by adding or removing absorbing areas from either side to bring about an acoustic stereo balance.

Referring to the diagram of our irregular room in Fig. 23, we can see that if we can find a line which bisects the room on an acoustically symmetrical basis, the stereo system can be placed accordingly. Then, as an extreme example, we can select another direction within the room which leaves the room unbalanced acoustically—and try to bring it into balance.

Referring to Fig. 24A, we have drawn such an acoustic dividing line from the corner casement window toward the dining area (there are other dividing lines which will be discussed in turn). It will be noted that there is fairly good geometric and acoustic symmetry about this line. There are two triangles of fairly equal acoustic influence on either side of this listening line. They have similar live walls opposite each other and then both end in an open area at the dining "L". We must now make a choice as to which end of the axis is to be used for listening and which end for speaker placement.

If the casement window area is used as the center of the settee and chair grouping, as shown in Fig. 24B, then the loudspeakers would be placed at the opposite end of the listening axis and, of course, on either side of it. It will be observed that the acoustic coverage of the listening area is fairly uniform from both sound sources. There are, however, some drawbacks to this arrangement other than the matter of decor!

Let us apply the five criteria for good stereo room adjustment discussed earlier. As far as (*a*), the room absorption characteristic, is concerned we can make the necessary adjustment irrespective of furniture placement. Concerning (*b*), the distribution of absorbent material, again this is relatively independent of furniture distribution. In relation to (*c*), the avoidance of large parallel reflecting areas, these again may be treated in a manner relatively independent of furniture placement. However, when we came to (*d*), the placement of the sound sources in comparatively live areas, the configuration of Fig. 24B is really "off base."

Both speakers are immediately adjacent to large open spaces at their sides and backs. This rather destroys any effective acoustic backing to reproject the sound into the listening area. This open area acts somewhat as an absorption area, thus the sound "throw" back into the room is stifled, rather than projected. We thus lose some of the liveness of the concert proscenium that we want to give the room the "concert hall" feeling.

To briefly examine the last criterion (*e*) where the listener should be in a generally "soft" acoustic location with some controlled liveness, this is readily taken care of by acoustic treatment involving drapes or acoustic tiles, in proper proportion to the bare plastered wall areas in the listening sector.

Let us now reverse the relative location of speakers and listeners in the room, putting the speakers near the corner area and the listeners at the open apex of the room, as shown in Fig. 24C. This latter suggestion may limit, or somewhat restrict, the number of permanently placed chairs but it does have certain acoustic advantages in keeping with the earlier enunciated principle of good listening room practice.

First, the speakers will be near a corner area of the room which not only enhances the lows but minimizes the creation of bothersome standing waves. Second, the corner itself may be kept fairly live due to the glass and plastered area behind the speakers; this gives a "stage" liveness to the sound source. Finally, the listener is in a rather open area where he is neither too close to excessive absorption which would deaden the sound nor yet too close to reflective surfaces which would cause excessive liveness directly at the ear. Despite these obvious acoustic advantages, the awkwardness of the seating arrangement and its limited stereo balanced area warrant exploration of still another room arrangement.

Decor May Unbalance Room

As far as the home decorator is concerned, the upper right-hand corner of the room (Fig. 24D) would seem to be a natural location for either a corner settee or lounge chairs and, without a doubt, such upholstered furniture will show up in this area. This area, then, may accommodate at least three to five persons in comfort. The central listening direction for this group will go diagonally across the room. The first thing we can do is put the speakers on either side of the dividing line and oriented towards the listening area.

Although we have physically balanced the listening area with the speakers, the room is far from being acoustically balanced on either side of the axis. Let us examine this unbalance and then correct it. On the left we have a plastered wall and a window section which produces a live condition. On the right we have a large open space leading off into the dining "L" which, because it is an open area, acts as an absorption area dependent on its own wall treatment. For the fortunate ones who live in southern climes, or even for northerners during the summer months, there is a quick and simple solution to this problem. If we simply open the casement windows in the corner we will have introduced considerable absorption into this live corner area, probably enough to equal the open area of the dining "L" on the opposite side of the listening axis. By this expedient, we may balance the room acoustically (provided the neighbors don't object). The preferred alternative is, of course, to acoustically "soften" this corner and "harden" the opposite open area.

Dealing first with the corner, we have a choice of draping the area and/or using acoustic tiles. The drapes will probably border the window and run from the ceiling molding to the floor. A spread of a couple of feet of drape on either side of the window will usually not suffice and probably acoustic tiles will have to be employed as well. Such tiles will not only help the acoustic situation but will add acoustic "finesse" to the room. Tiles placed above the window areas on both side and back walls and extending to the ceiling will remove much of the sound splash from the corners. The combination of acoustic tiles above the windows and the drapes on either side will go a long way toward softening the walls and corner.

If after making these adjustments to the corner wall area, we find we apparently hear more sound from the adjacent wall than from the opposite direction, then we will want to provide some reflective surface in that open area near the dining "L." For the modern homemaker a folding door across the dining "L" will not only enhance the appearance of the room but, when expanded across the area, provide a fair amount of acoustic reflective surface that will liven up that side of the room to balance the opposite corner.

As an alternative, a simple wood-panelled folding screen can be placed across the open area to add considerable acoustic "splash" and thus liven up the sound coming to the listening area from this section.

It would, of course, be impossible to outline all possible solutions to all room layouts. The examples just given, however, illustrate the general method of approach, first through determining the over-all acoustic condition of the room, then by adjusting it, when necessary, by the proper addition or removal of absorption material and, finally, by making symmetrical the *acoustic* condition of the room on either side of the listening axis.

The Coupled Room

We mentioned earlier that we would have something to say about "coupled areas." The problem, it will be recalled, came up during the initial discussion concerning the reverberation characteristics of the room of Fig. 23. In making the calculations we did not include the kitchen area on the grounds that it was a "coupled area." Now this simply means that, for acoustic absorption purposes, the surfaces of this room *cannot* be considered as direct extensions of the surfaces of the larger room. They are, in fact, not part of the main room but elements of a new room which, through the narrow archway or doorway, connects with or is coupled to the main room. The smaller room becomes an acoustic entity in itself—with its own reverberation characteristic and, more important, its own resonances. See Fig. 25.

The "coupled room" does make itself felt, however, not so much in terms of reverberation but in terms of reflected resonances. If a sound in the main room has a frequency component (in the low end of the audio spectrum) such that its wavelength is close to one of the dimensions of the smaller room, then this small room will be acoustically excited through the coupling door area, will then go into self resonance, and finally reflect the built up resonance condition back into the larger room.

The end effect of the "coupled" room will not be one of reverberation time but one of imposing a virtual resonant condition back into the listening room. The remedy for this is to see that any coupled room is itself treated so that no one axis of the room can be excited into resonance or to provide isolating panels or doors that will effectively destroy the common coupling area between the two rooms.

Once these stereo acoustic matters have been resolved, we are left with the one basic conflict that is as old as high-fidelity itself—the conflict between room decor and loudspeaker enclosure design. Shall we say (in a very small voice), Down Decor! Up Stereo?

Guest List

By CARL KOHLER

Portrayed here are a handful of guests any stereo-loving host may encounter . . .

KILLJOY

He seems to derive an almost sadistic satisfaction from finding minor flaws in his host's equipment and pointing out that the same turntables, tuners and amplifiers are selling for half of what the host paid, elsewhere. A genius at discovering minute damaged areas on cabinets which have escaped the attention of his host.

TINKERER

If not closely watched, he will quietly begin "investigating" any piece of hi-fi stereo equipment he can put his paws on. He can easily be spotted since he invariably carries a miniature tool kit in his hip pocket and is given to loitering around the backsides of cabinets.

CRITIC

This guest hasn't really approved of any recording since "Cohen On The Telephone" was issued. He listens to his host's latest and most unique records with utter dissatisfaction and frequent grunts of derision. Always giving the strong impression that he has been shanghaied into lending ears (which may be true), he has been known to stalk out during the first bars of music which keeps other people rooted to their chairs.

INDEX OF ADVERTISERS

The Audio Yearbook 1961 Edition

ADVERTISER:	PAGE NO.
Allied Radio	112
Ampex Magnetic Tape Products	1
Electronics Book Service	17, 44, 52, 53
Electronics Division Classified	35
Electronics Division Subscriptions	88
Electronics World Stereo Disc	87
Fidelitone	59
Fisher Radio Corporation	2nd Cover
Garrard Division of British Industries Corporation	4
Grommes Division of Precision Electronics, Inc.	21
Heath Company	36
Jobs and Careers in Electronics	73
Key Electronics Co.	83
North American Philips Co., Inc.	4th Cover
Olson Radio Corporation	83
Paco Electronics Co., Inc.	33
Photography Annual	98
RCA Institutes, Inc.	63
Schober Organ Corp., The	16
Scott, Inc., H. H.	11
Sherwood Electronic Laboratories, Inc.	23
Shure Brothers, Inc.	7
Stereo and Hi Fi Directory	103
Stromberg-Carlson Commercial Products Division	13
Technical Appliance Corporation	29
Thorens	3rd Cover
University Loudspeakers, Inc.	25

make a choice as to which end of the axis is to be used for listening and which end for speaker placement.

If the casement window area is used as the center of the settee and chair grouping, as shown in Fig. 24B, then the loudspeakers would be placed at the opposite end of the listening axis and, of course, on either side of it. It will be observed that the acoustic coverage of the listening area is fairly uniform from both sound sources. There are, however, some drawbacks to this arrangement other than the matter of decor!

Let us apply the five criteria for good stereo room adjustment discussed earlier. As far as (*a*), the room absorption characteristic, is concerned we can make the necessary adjustment irrespective of furniture placement. Concerning (*b*), the distribution of absorbent material, again this is relatively independent of furniture distribution. In relation to (*c*), the avoidance of large parallel reflecting areas, these again may be treated in a manner relatively independent of furniture placement. However, when we came to (*d*), the placement of the sound sources in comparatively live areas, the configuration of Fig. 24B is really "off base."

Both speakers are immediately adjacent to large open spaces at their sides and backs. This rather destroys any effective acoustic backing to reproject the sound into the listening area. This open area acts somewhat as an absorption area, thus the sound "throw" back into the room is stifled, rather than projected. We thus lose some of the liveness of the concert proscenium that we want to give the room the "concert hall" feeling.

To briefly examine the last criterion (*e*) where the listener should be in a generally "soft" acoustic location with some controlled liveness, this is readily taken care of by acoustic treatment involving drapes or acoustic tiles, in proper proportion to the bare plastered wall areas in the listening sector.

Let us now reverse the relative location of speakers and listeners in the room, putting the speakers near the corner area and the listeners at the open apex of the room, as shown in Fig. 24C. This latter suggestion may limit, or somewhat restrict, the number of permanently placed chairs but it does have certain acoustic advantages in keeping with the earlier enunciated principle of good listening room practice.

First, the speakers will be near a corner area of the room which not only enhances the lows but minimizes the creation of bothersome standing waves. Second, the corner itself may be kept fairly live due to the glass and plastered area behind the speakers; this gives a "stage" liveness to the sound source. Finally, the listener is in a rather open area where he is neither too close to excessive absorption which would deaden the sound nor yet too close to reflective surfaces which would cause excessive liveness directly at the ear. Despite these obvious acoustic advantages, the awkwardness of the seating arrangement and its limited stereo balanced area warrant exploration of still another room arrangement.

Decor May Unbalance Room

As far as the home decorator is concerned, the upper right-hand corner of the room (Fig. 24D) would seem to be a natural location for either a corner settee or lounge chairs and, without a doubt, such upholstered furniture will show up in this area. This area, then, may accommodate at least three to five persons in comfort. The central listening direction for this group will go diagonally across the room. The first thing we can do is put the speakers on either side of the dividing line and oriented towards the listening area.

Although we have physically balanced the listening area with the speakers, the room is far from being acoustically balanced on either side of the axis. Let us examine this unbalance and then correct it. On the left we have a plastered wall and a window section which produces a live condition. On the right we have a large open space leading off into the dining "L" which, because it is an open area, acts as an absorption area dependent on its own wall treatment. For the fortunate ones who live in southern climes, or even for northerners during the summer months, there is a quick and simple solution to this problem. If we simply open the casement windows in the corner we will have introduced considerable absorption into this live corner area, probably enough to equal the open area of the dining "L" on the opposite side of the listening axis. By this expedient, we may balance the room acoustically (provided the neighbors don't object). The preferred alternative is, of course, to acoustically "soften" this corner and "harden" the opposite open area.

Dealing first with the corner, we have a choice of draping the area and/or using acoustic tiles. The drapes will probably border the window and run from the ceiling molding to the floor. A spread of a couple of feet of drape on either side of the window will usually not suffice and probably acoustic tiles will have to be employed as well. Such tiles will not only help the acoustic situation but will add acoustic "finesse" to the room. Tiles placed above the window areas on both side and back walls and extending to the ceiling will remove much of the sound splash from the corners. The combination of acoustic tiles above the windows and the drapes on either side will go a long way toward softening the walls and corner.

If after making these adjustments to the corner wall area, we find we apparently hear more sound from the adjacent wall than from the opposite direction, then we will want to provide some reflective surface in that open area near the dining "L." For the modern homemaker a folding door across the dining "L" will not only enhance the appearance of the room but, when expanded across the area, provide a fair amount of acoustic reflective surface that will liven up that side of the room to balance the opposite corner.

As an alternative, a simple wood-panelled folding screen can be placed across the open area to add considerable acoustic "splash" and thus liven up the sound coming to the listening area from this section.

It would, of course, be impossible to outline all possible solutions to all room layouts. The examples just given, however, illustrate the general method of approach, first through determining the over-all acoustic condition of the room, then by adjusting it, when necessary, by the proper addition or removal of absorption material and, finally, by making symmetrical the *acoustic* condition of the room on either side of the listening axis.

The Coupled Room

We mentioned earlier that we would have something to say about "coupled areas." The problem, it will be recalled, came up during the initial discussion concerning the reverberation characteristics of the room of Fig. 23. In making the calculations we did not include the kitchen area on the grounds that it was a "coupled area." Now this simply means that, for acoustic absorption purposes, the surfaces of this room *cannot* be considered as direct extensions of the surfaces of the larger room. They are, in fact, not part of the main room but elements of a new room which, through the narrow archway or doorway, connects with or is coupled to the main room. The smaller room becomes an acoustic entity in itself—with its own reverberation characteristic and, more important, its own resonances. See Fig. 25.

The "coupled room" does make itself felt, however, not so much in terms of reverberation but in terms of reflected resonances. If a sound in the main room has a frequency component (in the low end of the audio spectrum) such that its wavelength is close to one of the dimensions of the smaller room, then this small room will be acoustically excited through the coupling door area, will then go into self resonance, and finally reflect the built up resonance condition back into the larger room.

The end effect of the "coupled" room will not be one of reverberation time but one of imposing a virtual resonant condition back into the listening room. The remedy for this is to see that any coupled room is itself treated so that no one axis of the room can be excited into resonance or to provide isolating panels or doors that will effectively destroy the common coupling area between the two rooms.

Once these stereo acoustic matters have been resolved, we are left with the one basic conflict that is as old as high-fidelity itself—the conflict between room decor and loudspeaker enclosure design. Shall we say (in a very small voice), Down Decor! Up Stereo?

Cable Address: ZIFFDAVIS

Chicago Los Angeles

ZIFF-DAVIS PUBLISHING COMPANY

One Park Avenue • New York 16, New York • Phone ORegon 9-7200

Electronics Annuals Division

Dear Audiophile

We hope that the AUDIO YEARBOOK-1961 has broadened your scope of understanding about the technical aspects of monaural and stereophonic sound recording and reproduction.

With your indulgence, we'd like to use the same medium, the AUDIO YEARBOOK-1961, to increase our understanding of you, the reader.

The following two pages contain a questionnaire which requests both general and specific information about you and your interest in audio. We'd very much like to have this information about you to help us in our preparation of future editions of AUDIO YEARBOOK.

Thanks in advance for your help, we really appreciate it.

Sincerely,

Tom Kneitel

Thomas S. Kneitel
Managing Editor

◀AUDIO YEARBOOK 1961▶

Reader's Confidential Report To The Editors

NAME ____________________

STREET & ADDRESS ____________________

CITY ____________ ZONE ______ STATE ____________

ABOUT THIS ANNUAL:

1. Where did you obtain it?
   Newsstand ☐ Parts Supplier ☐ Hi-Fi Dealer ☐
   From a friend ☐ Other (Please specify) ____________
2. Which electronics magazines do you read regularly? ____________
3. Have you ever read Audio Yearbook (formerly Hi-Fi Annual and Audio Handbook) before?
   Yes ☐ No ☐
4. How long will you keep this annual? ____________
5. How many other interested people will read or use your copy? ____________
6. Have you read any other Ziff-Davis Electronics Annuals? ____________
7. Which ones? ____________

ABOUT YOUR HI-FI INTERESTS:

8. a) How long have you been interested in Hi-Fi? ____________
   b) Do you (or does anyone in your household) own a complete, self-contained Hi-Fi set built by one manufacturer to be sold as a single unit (Package equipment)?
   Yes ☐ No ☐ (If "NO". . . skip to Q. 9a)
   c) If "YES": Please tell us how many sets you own, the type (stereo or monaural), the make and cost of each, and when it was purchased.
   Number owned: ____________

SET	MONAURAL	STEREO	MAKE	COST	YEAR BOUGHT
1	☐	☐	______	$______	______
2	☐	☐	______	$______	______
3	☐	☐	______	$______	______

   d) Please indicate where you purchased your package Hi-Fi set or sets.
   Appliance Store ☐ Music Store ☐
   Discount House ☐ Hi-Fi Salon ☐
   Department Store ☐ Other ____________ (Write in)

9. a) Do you own component, Hi-Fi equipment? (One or more components such as amplifier, tone arm, speaker, etc., which can be bought separately).
   Yes ☐ No ☐ (If "NO". . . skip to Q. 10a)
   b) If "YES": Please indicate below which components you own, the *number* you own, the number that were assembled from kits, and the total cost of each type of component owned.

	NOW OWN	NUMBER OWNED	NUMBER ASSEMBLED FROM KITS	TOTAL COST
Turntable (not record changer)	☐	______	______	$______
Tone Arm (for turntable only)	☐	______	______	$______
Record Changer and/or Manual Player	☐	______	______	$______
Tuner	☐	______	______	$______
Pre-Amplifier	☐	______	______	$______
Power Amplifier	☐	______	______	$______
Integrated Amplifier and Pre-Amplifier	☐	______	______	$______
Speaker System	☐	______	______	$______
Tape Recording or Playback System	☐	______	______	$______
Microphone	☐	______		$______
Phono Cartridge	☐	______		$______

   c) When you bought your last component, did you fill out a guarantee or warranty card and send it back to the manufacturer?
   Yes ☐ No ☐

10. a) Do you plan to buy any Hi-Fi components in the next 12 months?
   Yes ☐ No ☐ (If "No". . . skip to Q. 11a)
   b) If "YES": Please indicate below which components you will buy, the number you will buy, the number you will assemble from kits and how much you will expect to spend for each type of item.

	PLAN TO BUY	TOTAL NUMBER	NUMBER WILL ASEMBLE FROM KITS	ESTIMATED TOTAL COST
Turntable (not record changer)	☐	______	______	$______
Tone Arm (for turntable only)	☐	______	______	$______
Record Changer and or Manual Player	☐	______	______	$______
Tuner	☐	______	______	$______
Pre-Amplifier	☐	______	______	$______
Power Amplifier	☐	______	______	$______
Integrated Amplifier and Pre-Amplifier	☐	______	______	$______
Speaker System	☐	______	______	$______
Tape recording or playback system	☐	______	______	$______
Microphone	☐	______		$______
Phono Cartridge	☐	______		$______

11. a) Have you bought a replacement stylus (needle) in the past 12 months?
Yes ☐ No ☐ (If "NO": . . . skip to Q. 5a)
b) If "YES": Did you specify the brand of stylus wanted?
Yes ☐ No ☐ (Please skip to Q. 15a)
c) What brand did you buy?______

12. a) How much money did you spend in the last 12 months on Hi-Fi (*not* counting records or tapes)?
$______ None ☐
b) How many of each of the following items did you or your family buy during the past year?
Records______ None ☐
Pre-Recorded Tapes______ None ☐
Reels of raw Tape None ☐

ABOUT YOUR ELECTRONICS INTERESTS:

13. Do you have an electronics hobby such as: building electronics equipment from plans, servicing radio, TV, Hi-Fi or other electronics equipment for enjoyment *only* (not assembling Hi-Fi from major finished components or kits)?
Yes ☐ No ☐

14. a) Do you ever buy separate parts – for your business or hobby–such as tubes, capacitors, resistors, transistors, etc. for radio, TV, Hi-Fi or other electronics equipment?
Yes ☐ No ☐ (Please skip to Q. 10a.)
b) If "YES": Please tell us where you buy your electronics parts and about how often you buy from each source:
Local Radio-TV Service Shop or Dealer ☐ About______times a year.
Parts Jobber or Distributor (*not* mail order) ☐ About______times a year.
Mail Order ☐ About______times a year.
Other (please specify) ☐ About______times a year.

15. a) Do you derive *any* part of your income from selling or servicing radio, TV, Hi-Fi or other electronics equipment either full-time or part-time?
Yes ☐ No ☐ (Please skip to Q. 11a)
b) If "YES": Full-time ☐ or Part-time ☐

ABOUT YOURSELF:

16. Your approximate age, please?______Are you: Male ☐ Female ☐

17. a) Are you now employed?______either in business for yourself, or working for someone else?
Full-time ☐ Part-time ☐
Not presently employed (student, housewife, etc. do not count as employment) ☐
b) If now employed: What sort of work do you do? (please be specific).
______
______

18. Will you please check your approximate total family income bracket? (This is STRICTLY confidential.)

Less than $3,000 ☐
$3,000-$4,999 ☐
$5,000-$7,499 ☐
$7,500-$9,999 ☐
$10,000-$14,999 ☐
$15,000 and over ☐

(BE SURE YOUR NAME AND ADDRESS ARE PRINTED CLEARLY ON THE FIRST PAGE OF THIS QUESTIONNAIRE!)

FREE AIDS FOR AUDIOPHILES

● A library of useful information is available free, or at little cost thanks to the instructional literature many manufacturers offer. There are, no doubt, several booklets described here that will interest you. Just circle your choices on the coupon below and mail directly to AUDIO YEARBOOK 1961. We'll be happy to pass your requests to the manufacturers concerned.

1. Fisher Radio—Complete catalogue of Hi-Fi components and separate complete specification sheets.
2. Key Electronics—catalogue of all Hi-Fi equipment, audio components and Audio Catalogue A-12.
3. Norelco—Hi-Fi components Catalogue NAP 160.
4. RCA Institutes—Catalogue of basic and advanced electronics courses for home study.
5. Schober—16 page booklet about Schober Organ Kits. Also special brochure explaining operation of Schober Auto-Tuner for electronic organs.
6. Paco—Comprehensive catalogue of hi-fi components and test equipment, both kit and factory-wired.
7. Allied Radio (Knight)—"This Is Stereo High Fidelity", a 35 page color booklet about stereo. Explains frequency range, tone balance, description of all components. Also Allied's 1961 catalogue of electronic equipment and components including hi-fi furniture.
8. Fidelitone—Report and complete description of newly-designed Pyramid Stylus.
9. H. H. Scott—"How To Use Stereo High Fidelity Components In Your Decorating Plans", and booklet describing Stereomaster components. Separate specification sheets also available.
10. Stromberg Carlson—48 page brochure on component ensembles. Also catalogue and specifications for tuners, turntables, amplifiers.
11. Thorens — Complete specifications on turntable kits, including laboratory reports.
12. British Industries—Comparator sheets with complete specifications and illustrations of RJ Enclosures, Garrard Record Changers, Wharfedale Speakers, Leak Amplifiers, Multicore Solder.
13. Sherwood—Complete specification sheets on all Sherwood hi-fi components.
14. Shure Bros.—Two booklets, "Art of Selecting, Playing and Preserving Recordings", 25¢, and "The Critical Ear", a discussion of high-fidelity.
15. Taco—Technical brochure on FM Antennas, "Why An FM Antenna"
16. Heath—76 page catalogue of hi-fi components, marine, ham, test equipment, and products for the home.
17. Olson Radio—Catalogue issued every six weeks includes hi-fi components, test equipment, accessories, and radios.
18. Ampex Magnetic Tape—Complete specification sheets on magnetic tapes. Also two informative brochures, "How To Make Better Tape Recordings" and "How To Tape Off The Air".
19. University Loudspeakers, Inc.—"Hi-Fi – An Informative Guide to Stereo And Monophonic Speaker Systems and Components."
20. Precision Electronics—Full color catalogue of current Grommes line along with detailed specifications.

Coupon valid until March 31, 1961, after that date make your selections from ELECTRONICS WORLD monthly listing of free literature.

USE THIS COUPON FOR FREE LITERATURE

To: AUDIO YEARBOOK-1961
P. O. BOX 209
VILLAGE STATION
N. Y. 14, N. Y.

Please arrange to have the booklets whose numbers I have encircled sent to me as soon as possible.

1 2 3 4 5 6 7 8 9 10

11 12 13 14 15 16 17 18 19 20

Total number of booklets requested.

$________Total Amount enclosed for pay booklets.

NAME________________________________

ADDRESS________________________________

CITY________________ZONE______STATE________________

Guest List

By CARL KOHLER

Portrayed here are a handful of guests any stereo-loving host may encounter . . .

KILLJOY

He seems to derive an almost sadistic satisfaction from finding minor flaws in his host's equipment and pointing out that the same turntables, tuners and amplifiers are selling for half of what the host paid, elsewhere. A genius at discovering minute damaged areas on cabinets which have escaped the attention of his host.

TINKERER

If not closely watched, he will quietly begin "investigating" any piece of hi-fi stereo equipment he can put his paws on. He can easily be spotted since he invariably carries a miniature tool kit in his hip pocket and is given to loitering around the backsides of cabinets.

CRITIC

This guest hasn't really approved of any recording since "Cohen On The Telephone" was issued. He listens to his host's latest and most unique records with utter dissatisfaction and frequent grunts of derision. Always giving the strong impression that he has been shanghaied into lending ears (which may be true), he has been known to stalk out during the first bars of music which keeps other people rooted to their chairs.

INDEX OF ADVERTISERS

The Audio Yearbook
1961 Edition

ADVERTISER:	PAGE NO.
Allied Radio	112
Ampex Magnetic Tape Products	1
Electronics Book Service	17, 44, 52, 53
Electronics Division Classified	35
Electronics Division Subscriptions	88
Electronics World Stereo Disc	87
Fidelitone	59
Fisher Radio Corporation	2nd Cover
Garrard Division of British Industries Corporation	4
Grommes Division of Precision Electronics, Inc.	21
Heath Company	36
Jobs and Careers in Electronics	73
Key Electronics Co.	83
North American Philips Co., Inc.	4th Cover
Olson Radio Corporation	83
Paco Electronics Co., Inc.	33
Photography Annual	98
RCA Institutes, Inc.	63
Schober Organ Corp., The	16
Scott, Inc., H. H.	11
Sherwood Electronic Laboratories, Inc.	23
Shure Brothers, Inc.	7
Stereo and Hi Fi Directory	103
Stromberg-Carlson Commercial Products Division	13
Technical Appliance Corporation	29
Thorens	3rd Cover
University Loudspeakers, Inc.	25

The greatest beauty of Thorens famous quality... *you can afford it!*

No need to hesitate, you *can* afford Thorens famous quality. You *can* have music as it's meant to be heard. You *can* relax with Thorens *unique* one year guarantee. There's a Thorens model that fits handily into any budget.

Whether you know a lot or a little about high-fidelity equipment, you'll particularly enjoy the courteous and knowledgeable way a Thorens franchised dealer *earns* your confidence. Each Thorens dealer is carefully selected for knowledge, ability and integrity. They'll make buying *your* Thorens almost as much fun as owning it. Shop around this page for a few of the outstanding features and then stop in and *see* all of them for yourself. O.5

Guaranteed for one full year. Sold only through carefully selected franchised dealers.

THORENS FULL YEAR GUARANTEE SWISS MADE PRODUCTS

THORENS

SWISS MADE PRODUCTS
MUSIC BOXES • HI-FI COMPONENTS
SPRING-POWERED SHAVERS • LIGHTERS
New Hyde Park, N.Y.

MATCHLESS!
TD-124. All four speeds. Plays any record. Easy-to-use lighted strobe sets exact speed for best musical reproduction. Completely silent. Many more exclusive features... only $99.95 net.

TD-184

TD-124

MORE ECONOMICAL!
TD-134. The finest 4-speed manual turntable you can buy. Includes tone arm. Elimination of semi-automatic feature saves you another $15. You can also save up to $30 on the tone arm. Look at TD-134 . . . only $59.95 net.

MARVELOUS!
TD-184. Includes tone arm and simple dialing system that lets you select records and start turntable. All 4 speeds. Save $20 on turntable, up to $30 on tone arm. Look at TD-184 . . . only $75.00 net.

TD-134

TDK-101

MOST ECONOMICAL!
TDK-101. You can assemble this Thorens turntable yourself. The superb quality of the components makes all your work worthwhile. Look at Thorens TDK-101 . . . only $47.50 net.

guild-crafted by Philips of the Netherlands to give you

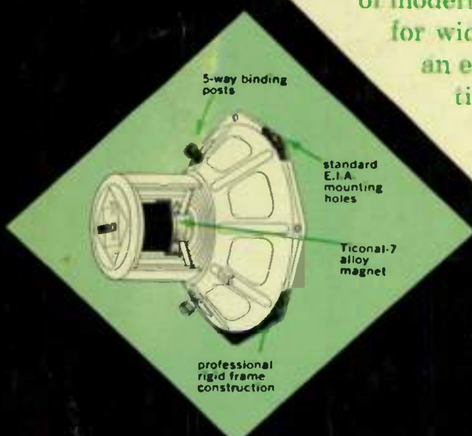
THE CLEANEST SOUND AROUND...

Norelco®

T-7 LOUDSPEAKERS

featuring voice coil magnets of TICONAL VII
(30% more powerful than Alnico V)

This new line of 5" to 12" loudspeakers is designed to match the quality requirements of the discriminating music lover . . . at surprisingly moderate prices. The world's greatest buys on the basis of listening quality, the T-7 series incorporates voice coil magnets of Ticonal-7 steel, the most powerful of modern magnet alloys, for maximum efficiency... dual cones for wide frequency response . . . constant impedance, for an extremely straight response curve... longer effective air-gaps and extra high flux density to eliminate ringing and overshoot... everything needed to guarantee completely unadulterated audio realism.



Norelco ®

EXHIBITION SPEAKER ENCLOSURES

Acoustically designed to achieve moderate size without loss of bass response or wideness of dispersion. Perfectly matched for bass response of the NORELCO T-7 speakers . . . ideal for most quality 8" to 12" speakers. Available in three sizes, in hand-rubbed Mahogany, Walnut, Blond or Cherry.

Norelco

STEREO/TURNTABLE RECORD CHANGER

A versatile, low-cost record changer offering years of trouble-free stereo and monophonic reproduction . . . interchangeable turntable/changer spindles . . . automatic, semi-automatic and manual play... 4 speeds... automatic 7", 10", and 12" intermix.

Norelco

CONTINENTAL "400" TAPE RECORDER (EL3536/54)

◆4-track stereophonic or monophonic recording and playback ◆ 3 tape speeds ◆ completely self-contained, including dual recording and playback preamps, dual power amplifiers, 2 Norelco wide-range speakers (2nd in lid), stereodynamic microphone.

Norelco

CONTINENTAL "300" TAPE RECORDER (EL3542)

◆ 4-track stereo playback ◆ 4-track mono record-playback ◆ 3-speeds ◆ completely self-contained ◆ mixing facilities ◆ 'sound-on-sound' button ◆ suitable for use as a phono/P.A. system ◆ can also play through external hi-fi system ◆ dynamic microphone.

Norelco

CONTINENTAL "200" TAPE RECORDER (EL3541)

◆ single-speed (7½ ips) 4-track stereo playback, 4-track mono record-playback ◆ completely self-contained, including microphone ◆ mixing facilities ◆ 'sound-on-sound' button ◆ can also play through external system ◆ self-contained phono/P.A. system.

For additional descriptive literature, write to:
NORTH AMERICAN PHILIPS CO., INC.
High Fidelity Products Division
230 Duffy Avenue, Hicksville, L. I., New York