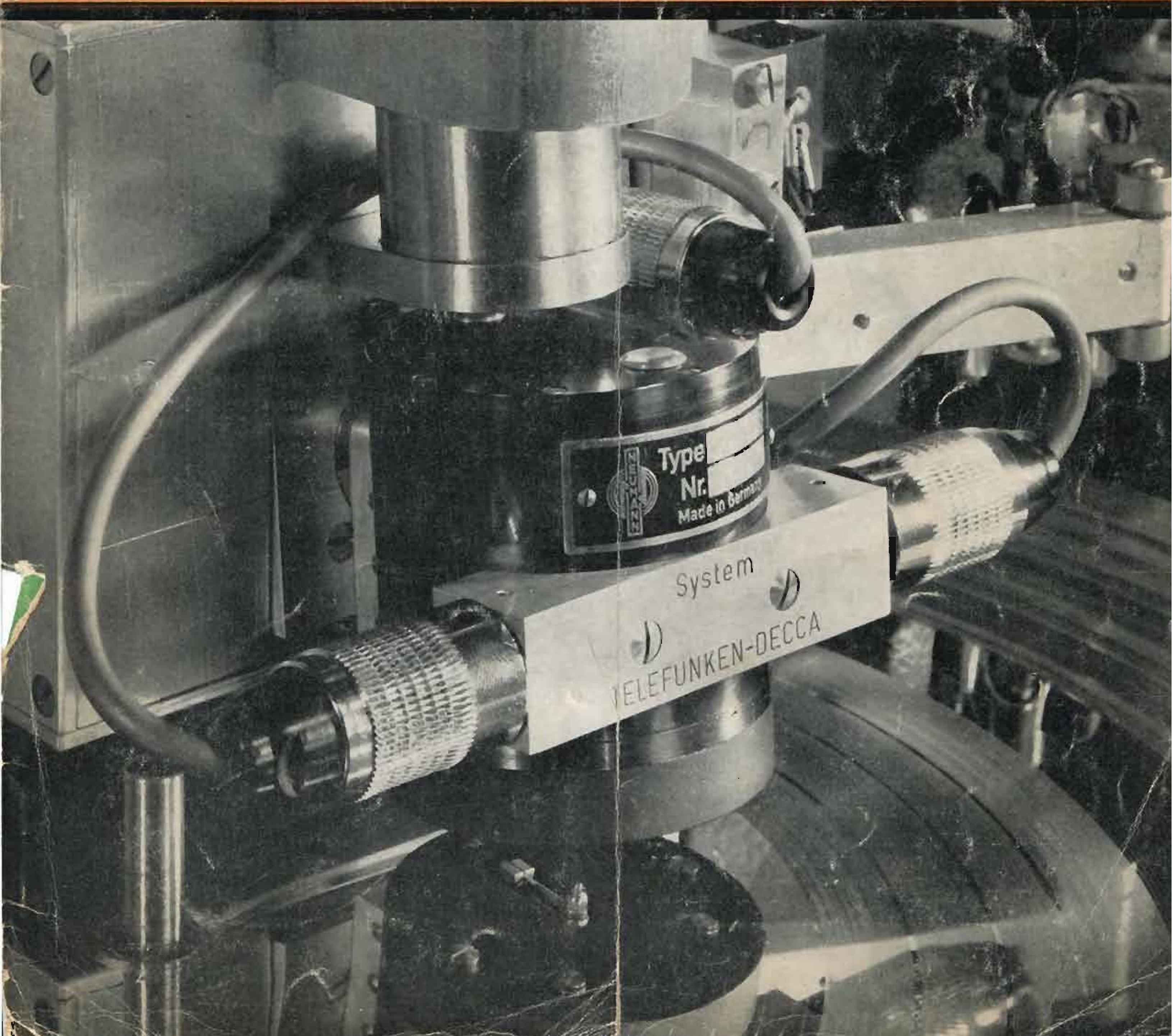
AUDIO

NOVEMBER, 1958
50¢

...the original magazine about high fidelity!



1955

The recording tape industry introduces the new "double play" tapes, made on ½-mil Mylar* polyester film base, making available twice the normal length of tape on any given reel size and effectively doubling the normal playing time. *Problem:* The new tape is "twice as long," to be sure, but quite fragile, requiring special care in handling.

1957

The recording tape industry introduces the new "tensilized" or "fortified" double play tapes, now made on a special type of reinforced ½-mil Mylar* base that is twice as resistant to stretching and breaking as in the 1955 kind. *Problem:* The new tape is indeed "twice as long and twice as strong" now (just as strong as normal tape, in fact), but the price is astronomical.

1958

Beginning November 1, 1958, all **irish** "Double Play" recording tape on the market will be of the reinforced, 1957 kind — *but* at the moderate price of the older, 1955 kind. *End of Problem:* This latest **irish** "Double Play" tape has the *length* (2400 feet on a standard 7-inch reel), it has the *strength* (6 lbs. tensile force) — and you can afford it!

*DuPont's registered trademark

NOVEMBER, 1958 VOL. 42, No. 11
Successor to RADIO, Est. 1917.

AUDIO

ENGINEERING MUSIC SOUND REPRODUCTION

CONTENTS

COVER PHOTO—The Neumann-Teldec ZS 90/45 stereo cutterhead shown mounted in its automatic suspension on the Neumann stereo mastering lathe. This cutterhead, described in the article commencing on page 26, has been in use for the past 18 months by both Telefunken and English Decca (London) records in the cutting of stereo releases.

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AUDIOCLINIC??

JOSEPH GIOVANELLI*

Rotating Speaker Speed Control

Q. *I have built a rotating speaker system to be used as part of the tremolo effect with my organ. My problem is that of controlling the speed of rotation in order to obtain varying degrees of tremolo. A rectifier and d.c. motor would be the answer, but I do not know how to design this type of rectifier circuit. Can you tell me how to design a suitable circuit? Robert McDonald, Oakland, Calif.*

A. I recommend the standard bridge-rectifier circuit for this purpose. The current handling capacity of the individual diodes making up the bridge should be somewhat larger than the d.c. drain of the motor. You will need between 200 and 1000 µf, of filtering. 500 mils requires about 200 µf, and the amount of filtering increases with the amount of current taken by the motor. Insert a 5 or 10 ohm resistor in series with the filter bank and the rectifier to prevent surge current from damaging the rectifier, and overheating the filter bank. The heating effect upon the capacitors can be further minimized by using several small capacitors rather than a single large unit. The actual mechanism for controlling the speed can be either a series rheostat of appropriate resistance and wattage or a Variac.

An easier solution to your problem might lie in the use of a universal a.c.-d.c. motor. Again you can either use a rheostat or Variac as the agent for controlling the speed. With either type of motor, it is a good idea to bypass each brush to ground in order to minimize hash interference.

Distortion and Volume Controls

Q. *My power amplifier is behaving in a most peculiar manner. When the volume control is turned up full, the bass response is normal, but as the setting of the control is reduced, the bass falls off sharply. This is not a result of the Fletcher-Munsen effect, for, as I advance the gain of preamplifier at the same time that I decrease that of the power amplifier, the bass still continues to fall off. Enclosed is a schematic of the input circuit of the power amplifier,* (Fig. 1), *perhaps it will help you to determine what is wrong. James P. Gooley, Oak Lawn, R. I.*

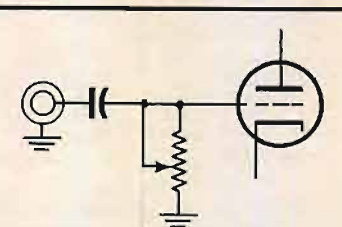


Fig. 1

A. Notice that in the input circuit of your amplifier the volume control is connected as a rheostat. This circuit can perform the function of reducing the gain of the amplifier, but, as you have seen, the bass will be attenuated in greater amount than the remainder of the signal as the gain is lowered.

To explain why this circuit causes this behavior, let us assume that the reactance of the coupling capacitor is 0.5 megohm at 30 cps, and that the resistance value of the potentiometer when fully open is also 0.5 megohm. At 30 cps, half the voltage developed by the preamplifier will be lost across the coupling capacitor, while the remaining half is available for application to the grid of the input tube of the power amplifier. Let us assume that the gain has been reduced, and the resistance of the control is now 0.25 megohm. Of course the over gain of the amplifier has been reduced 6 db, but our 30-cps tone has been decreased by an even greater amount. The reactance of the capacitor is still 0.5 megohm, but the resistance of the pot has been reduced so now only ⅓ the voltage produced by the preamplifier is available to the power amplifier. This effect will become more and more severe as the resistance of the potentiometer decreases. What is needed is a control circuit which maintains a constant reactance, but still allows the grid to pick off as much signal as is needed, by means of voltage divider action. *Figure* 2 shows how this same 0.5-meg. control can be wired to accomplish this. Note that the coupling capacitor is connected through the full resistance of the pot to ground. The slider has no effect upon the reactance presented to the coupling capacitor, since the grid to which it is attached draws no current and is therefore of infinite resistance.

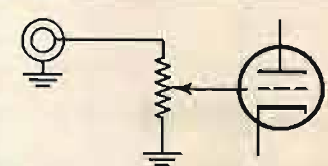


Fig. 2

Speakers and Infinite Baffles

Q. *We are considering the installation of an infinite baffle in a wall. Free cone resonance and efficiency are doubtless to be selected with some discretion in this regard. We understand that some of the best speakers for this use are in the relatively inefficient class. How is better bass definition attributed to low efficiency? What bearing has the flux density upon efficiency? What cone resonance will be best for most realistic reproduction in the home? L. B. Osborn, Newburgh, Indiana.*

A. Efficiency is governed by the flux density and compliance. The less the compliance and the greater the flux density, the greater the efficiency of the speaker. Because the back wave is lost with infinite baffles, the cone must be free to travel a great distance to make up for this loss.

* *3420 Newkirk Ave., Brooklyn 3, N. Y.*

British Industries Corporation Stereo Demonstrations

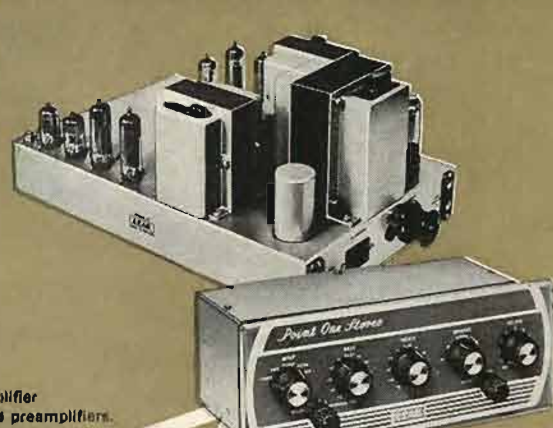
at high fidelity shows throughout the country...

1. The records were played on a

Garrard
WORLD'S FINEST RECORD CHANGER

We used the Garrard Model RC88 because this changer reproduces stereo music precisely as recorded, without introducing any disturbing or distorting factors such as rumble or wow. In fact, the Garrard changer performs better than most so-called "professional" turntables, because this changer is actually a superb turntable combined with a scientifically engineered aluminum tone arm which tracks at the correct stylus pressure without undesirable resonances. You have the added convenience of not only being able to play records simply by hand, but also stacked, with the assurance that they are being handled automatically more gently than by any human hand. This is true on a Garrard changer and at a price much lower than you may have been led to expect. When considering the claims of changers "specifically designed for stereo", or turntables "to play stereo properly", it is well to remember that for years Garrard changers have had all the qualities necessary for this type of sensitive reproduction. Now more than ever before, it is essential to insist on a Garrard changer, and accept no substitute, if you want the finest Stereo or, of course, Monaural reproduction.

The Garrard line includes changers, manual players and transcription turntables; six models: $32.50 to $89.00.



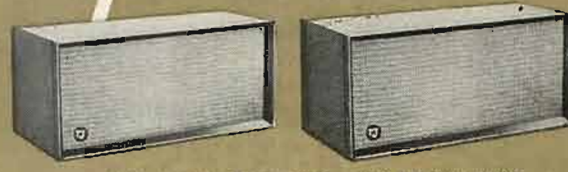
2. The new **LEAK** stereo amplifier and preamplifier were used to control the sound. Leak professional quality amplifiers are unique because they keep distortion down to a trivial one-tenth of one percent (0.1%) at full rated power... the lowest figure ever achieved! This explains why the Leak stereo sound you heard was so natural, and why you could have enjoyed it for hours without fatigue. Low distortion is the key to quality stereo reproduction. Invest wisely. It is important to remember that the amplifier and preamplifier are the very heart of your stereo system.

The Leak line includes two stereo amplifiers and a stereo preamplifier plus a complete group of monaural power amplifiers and preamplifiers.

3. You heard the **Wharfedale** SFB/3 sand-filled speaker systems... one "Custom" and one "Deluxe"... both models identical in speakers and sound but in different enclosures. This is an entirely new development in three-speaker systems by England's G. A. Briggs. You may have noticed how the absence of resonance and the clean, spacious sound, heightened the stereo effect.

The Wharfedale line includes full range, bass and treble speakers; two and three-way speaker systems and speaker enclosures.

4. A pair of **RJ** super-8's were the alternate speaker system. You may have wondered how such splendid sound could come from so small a speaker enclosure. The answer lies in patented R-J design principles, which mean that no other small enclosure can match the RJ in performance. Stereo does create some new problems in room arrangement which are easily solved with versatile R-J enclosures.



The R-J line includes 5 enclosures... shelf and floor models.

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We have prepared a series of Comparator Guides covering the various BIC product lines. If you have not already obtained them at the show, we will be happy to send them to you. Please write, specifying the BIC products which interest you, Dept. AS-18.

BRITISH INDUSTRIES CORPORATION, Port Washington, N. Y.

Since low compliance and high flux density both hinder cone travel, speakers having these characteristics will not work well in an infinite baffle. If the cone is too compliant for the amount of rear loading upon it, the speaker may be damaged. For this reason, I like to use a speaker with rather high compliance and high flux density. The flux density limits cone travel after the cone is in high-amplitude motion and it thereby prevents possible damage during power peaks. Poor compliance is a constant restraining force and should be avoided. Unless your baffle is specially designed, I don't recommend a speaker having a very high compliance. You will have lots of intermodulation distortion at best, and you can probably damage the speaker mechanism.

Since an infinite baffle tends to raise the resonant frequency of the speaker cone, it is advisable to use a speaker whose resonance point is as low as possible. As the compliance increases, resonance is automatically lowered. If the infinite baffle is located in a small room, it is unnecessary to have a speaker with a resonant frequency much below 30 cps, since the volume of air contained in the room is not sufficient to reproduce frequencies much below 35 cps. A speaker's 30-cps resonance may be raised easily to 35 cps or higher, depending upon the size of the space behind the cone.

Interference from TV receivers

Q. *When I tune my receiver through the AM band, I find it covered with a series of whistles whose pitch varies according to the station to which I am listening. Since this phenomenon occurs with many other receivers in the apartment house, I don't think it is because of any possible misadjustment of my whistle filter. The only time the whistles do not appear is early in the morning. When my TV set or someone else's, is turned on, the whistle reappears. What is causing this, and what can I do about it? Woodrow C. Doebler, Pottstown, Pa.*

A. This series of whistles is the result of beats between broadcast stations and harmonics of the horizontal oscillator of the offending TV receiver. Since the space between harmonics is small (the frequency of the horizontal oscillator being 15,750 cps), they can beat with any station.

There are two ways by which the harmonics can enter your receiver. The first of these is by direct radiation from the TV set, and the harmonics of the oscillator are picked directly by the receiver's antenna. This effect can sometimes be minimized by placing your antenna high enough so that it will be out of the radiation field. (Some receivers have been known to radiate several hundred feet, in which case you cannot erect an antenna high enough.) This can also result in the strength of the desired broadcast stations' being great enough to override the whistles. The antenna should be fed by coaxial cable, so that no TV set radiation will be picked up by the lead-in.

The second and less unlikely means of entry of these undesired signals is through the power lines. This may be overcome by inserting an r.f. choke in each side of the line feeding your tuner, and bypassing all choke leads to ground with good-quality mica capacitors. Æ

LETTERS

Reply to the Editor

SIR:

We would beg to differ with some aspects of your October EDITOR'S REVIEW.

WBAI-FM has been, since September 12th, stereocasting via the Crosby compatible multiplex system between the hours of 4–5 and 7–8 p.m. daily.

We chose the Crosby system for a variety of reasons—the two most important were the fact that if offered a fully compatible signal (A plus B) to the listener with a single FM set only, and that if offered 15-kc response on both the main and the subcarrier channels. It was, we felt, the only device available consistent with the obligation a high-fidelity broadcasting station owed its listeners. If we were to broadcast via any device which offers less fidelity than that which FM itself is capable of, this would not, we feel, be serving the public interest, necessity, and convenience to the best of our ability. If we were to compromise with lesser fidelity we would not need Frequency Modulation itself at all. If a higher fidelity system is evolved, WBAI will be the first to use it.

The growing public interest in high fidelity and stereo imposes an obligation upon the broadcaster which we feel heavily. As our audience grows it is even more contingent upon us to provide it with the best that is technologically feasible. Less than the best which is available is emphatically NOT good enough. To take the attitude, as has been several times expressed recently, that less than that which we are capable of will satisfy the public is, we feel, an evasion of responsibility and duty. The public can and does know the difference between full 15-kc fidelity (with its delicate harmonics and overtones) and less. If it did not, it would not be investing ever-increasing amounts of money in equipment for high fidelity disc and tape and broadcast reproduction and reception.

The editorial we refer to also says "the presence of wide-band multiplex adapters in the hands of the public would certainly lower the value of the background music services, even though 'pirating' of the programs is illegal. There are always some who risk the illegality for the sake of a few dollars."

We submit that this charge of piracy, raised by certain broadcasters in the first place, is in effect a large red herring. So charging the public, at this stage of the art of multiplex's development, is equivalent to saying that a man may, someday, conspire to commit a crime. We contend that any such judgment should be made after the fact and not before. It is our feeling that one does not accuse the public before the act has been committed. Certainly, if nothing else, such accusation runs directly counter to the old American tradition that the accused is innocent until *proven* guilty. It is possible that, some day, someone may commit the crime we refer to, but this is primarily a police problem, not a communications one, and to raise it in this particular context, as an excuse for not providing the public with the best to which it is entitled, is raising a false issue indeed.

We do not, actually, intend to quarrel with the broadcaster who wishes to utilize more than one multiplex channel (at less than 15 kc) as an aid to supplementing the income his station can provide. *We only request that he does not attempt to provide the public with less than the best that can be done—and call it "as good"—in an attempt to cash in on the public's interest in stereophonic broadcasting.* Better than 7½-kc response is technologically feasible, and available. Let us not confuse motives at this stage in the development of the art of high-fidelity broadcasting.

Less than the best the broadcaster can provide to his audience cannot be "as good" as the best. It is our intention to do anything we can to further the growth of high-fidelity broadcasting, utilizing the best means available, as a measure of our responsibility to the public's interest, necessity, and convenience.

BERT COWLAN, General Manager.
Radio Station WBAI,
2 E. 61st St.,
New York 21, N. Y.

SIR:

Your October editorial indicates that you favor a system of FM multiplex stereo which allows for a channel of background music or other point-to-point service instead of the single-channel wide-band system which I propose, but I am convinced that you formed your opinion with a lack of engineering data. I have accumulated a large amount of data which I intend to publish in detail, but I would like to give a brief resume of it in the hopes that it will provide clarification of some of the points in your editorial.

Your first point is a fear of "piracy" of background music. I would like to point out that the wide-band type of stereo adapter is not a proper receiver for this background music.[1] The transmission of this music utilizes a technique of "muting" between musical numbers such that the stereo listener receives an extremely uncomfortable blast of noise approximately 12 db louder than the music during the muting period. The restaurant subscribing to the service is equipped with a special receiver which automatically mutes out this blast of noise. Practically all of the background music stations use this technique. If they do not, commercial equipment for doing so is available to them.

From an engineering standpoint, it appears possible, at least on paper, to design a system which will allow a background music subcarrier, but there are sacrifices which I feel your editorial passed over too lightly. The most important first consideration is that of the power loss of the station considered as a stereo broadcasting station. The stereo transmission is completely dependent on the transmission efficiency of the subcarrier. The inherent transmission efficiency of any subcarrier modulation system is always poorer than that of the main channel. This is especially true in the case of frequency modulation which has a triangular noise spectrum such that the received noise rises in direct proportion to the modulating frequency so that a high modulating frequency in the subcarrier region is in a high level of noise. For the standards I have proposed, namely a 25-kc deviation applied to a 50-kc subcarrier applied at 50 per cent modulation, the difference in signal-to-noise ratio between the main channel and the subcarrier channel is approximately 12 db. This figure

(Continued on page 70)

[1] *Multiplex adapters can be made that do silence upon absence of carrier. Refer back to Mr. Day's article in the August issue.* ED.

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MODEL SE-1 (center unit) $149.95
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Superbly designed cabinetry to house your complete stereo system. Delivered with pre-cut panels to fit Heathkit AM-FM tuner (PT-1), stereo preamplifier (SP-1 & 2) and record changer (RP-3). Blank panels also supplied to cut out for any other equipment you may now own. Adequate space also provided for tape deck, speakers, record storage and amplifiers. Speaker wings will hold Heathkit SS-2 or other speaker units of similar size. Available in unfinished birch or mahogany plywood.

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HIGH FIDELITY RECORD CHANGER KIT

MODEL RP-3 $64.95

Turntable quality with fully automatic features! A unique "turntable pause" allows record to fall gently into place while turntable is stopped. The tone arm engages the motionless record, and a friction clutch assures smooth start. Automatic speed selector plays mixed 33⅓ and 45 RPM records regardless of sequence. Four speeds available: 16, 33⅓, 45 and 78 RPM. Changer complete with GE-VR-II cartridge with diamond LP and sapphire 78 stylus, changer base, stylus pressure gauge and 45 RPM spindle. Shpg. Wt. 19 lbs.

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The 10-tube FM circuit features AFC (automatic frequency control) as well as AGC. An accurate tuning meter operates on both AM and FM while a 3-position switch selects meter functions without disturbing stereo or monaural listening. Individual flywheel tuning on both AM and FM. FM sensitivity is three microvolts for 30 db of quieting. The 3-tube FM front end is prewired and prealigned, and the entire AM circuit is on one printed circuit board for ease of construction. Shpg. Wt. 20 lbs.

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A real work horse packed with top quality features, this hi-fi amplifier represents a remarkable value at less than a dollar per watt. Full audio output at maximum damping is a true 55 watts from 20 CPS to 20 kc with less than 2% total harmonic distortion throughout the entire range. Featuring famous "bas-bal" circuit, push-pull EL34 tubes and new modern styling. Shpg. Wt. 28 lbs.

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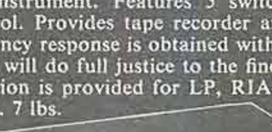
All the controls you need to master a complete high fidelity system are incorporated in this versatile instrument. Features 5 switch-selected inputs each with level control. Provides tape recorder and cathode-follower outputs. Full frequency response is obtained within ±1½ db from 15 to 35,000 CPS and will do full justice to the finest available program sources. Equalization is provided for LP, RIAA, AES, and early 78 records. Shpg. Wt. 7 lbs.



HIGH FIDELITY TAPE RECORDER KIT

MODEL TR-1A **$99.95**

Includes tape deck assembly, preamplifier and roll of tape.

MODEL TE-1 **$39.95**

Shpg. Wt. 10 lbs. (Tape Preamplifier Only)

The model TR-1A provides monaural record/playback with fast forward and rewind functions. 7½ and 3¾ IPS tape speeds are selected by changing belt drive. Flutter and wow are held to less than 0.35%. Frequency response at 7½ IPS ±2.0 db 50-10,000 CPS, at 3¾ IPS ±2.0 db 50-6,500 CPS. The model TE-1 record/playback tape preamplifier, supplied with the mechanical assembly, provides NARTB playback equalization. A two-position selector switch provides for mike or line input. Separate record and playback gain controls. Cathode follower output. Complete instructions provided for easy assembly. Signal-to-noise ratio is better than 45 db below normal recording level with less than 1% total harmonic distortion. (Tape mechanism not sold separately). Shpg. Wt. 24 lbs.

HIGH FIDELITY AM TUNER KIT

MODEL BC-1A **$26.95**

Designed especially for high fidelity applications this AM tuner will give you reception close to FM. A special detector is incorporated and the IF circuits are "broadbanded" for low signal distortion. Sensitivity and selectivity are excellent and quiet performance is assured by a high signal-to-noise ratio. All tunable components are prealigned before shipment. Your "best buy" in an AM tuner. Shpg. Wt. 9 lbs.

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MODEL FM-3A **$26.95**

For noise and static-free sound reception, this FM tuner is your least expensive source of high fidelity material. Efficient circuit design features stabilized oscillator circuit and broadband IF circuits for full fidelity with high sensitivity. All tunable components are prealigned before shipment. Edge-illuminated slide rule dial. Covers complete FM band from 88 to 108 mc. Shpg. Wt. 8 lbs.

"UNIVERSAL" 12 WATT AMPLIFIER KIT

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Ideal for stereo or monaural applications, this 12-watt power package features less than 2% total harmonic distortion throughout the entire audio range (20 to 20,000 CPS) at full 12-watt output. Use with preamplifier models WA-P2 or SP-1 & 2. Taps for 4, 8 and 16 ohm speakers. Shpg. Wt. 13 lbs.

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Control your complete home hi-fi system right from your easy chair with this handsome chairside enclosure in either traditional or contemporary models. It is designed to house the Heathkit AM and FM tuners (BC-1A and FM-3A) and the WA-P2 preamplifier, along with the RP-3 or majority of record changers which will fit in the space provided. Well ventilated space is provided in the rear of the enclosure for any of the Heathkit amplifiers designed to operate with the WA-P2. The tilt-out shelf can be installed on either right or left side as desired during the construction, and the lift-top lid in front can also be reversed. All parts are precut and predrilled for easy assembly. The contemporary cabinet is available in either mahogany or birch, and the traditional cabinet is available in mahogany suitable for the finish of your choice. All hardware supplied. Shpg. Wt. 46 lbs.

IT'S EASY . . . IT'S FUN AND YOU SAVE UP TO ½ WITH DO-IT-YOURSELF HEATHKITS

Putting together your own Heathkit can be one of the most exciting hobbies you ever enjoyed. Simple step-by-step instructions and large pictorial diagrams show you where every part goes. You can't possibly go wrong. No previous electronic or kit building experience is required. You'll learn a lot about your equipment as you build it, and, of course, you will experience the pride and satisfaction of having done it yourself.

"BASIC RANGE" HI-FI SPEAKER SYSTEM KIT

MODEL SS-2 **$39.95**

Legs: No. 91-26 Shpg. Wt. 3 lb. $4.95

The modest cost of this basic speaker system makes it a spectacular buy for any hi-fi enthusiast. Uses an 8" mid-range woofer and a compression-type tweeter to cover the frequency range of 50 to 12,000 CPS. Crossover circuit is built in with balance control. Impedance is 16 ohms. Power rating 25 watts. Tweeter horn rotates so that the speaker may be used in either an upright or horizontal position. Cabinet is made of veneer-surfaced furniture-grade plywood suitable for light or dark finish. All wood parts are precut and predrilled for easy assembly. Shpg. Wt. 26 lbs.

DIAMOND STYLUS HI-FI PICKUP CARTRIDGE

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Replace your present pickup with the MF-1 and enjoy the fullest fidelity your library of LP's has to offer. Designed to Heath specifications to offer you one of the finest cartridges available today. Nominally flat response from 20 to 20,000 CPS. Shpg. Wt. 1 lb.

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The SS-1B employs a 15" woofer and super tweeter to extend overall response of basic SS-2 speaker from 35 to 16,000 CPS ±5 db. Crossover circuit is built in. Impedance is 16 ohms, power rating 35 watts. Constructed of ¾" veneer-surfaced plywood suitable for light or dark finish. Shpg. Wt. 80 lbs.

MODEL SS-1B
$99.95

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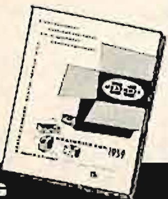

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STEREO SALAD

My intense preoccupation with stereo listening, inevitable during these last months, has played hob with more than plain mono listening though I have heaps of perfectly good non-stereo discs that I somehow can't ever seem to get around to. Hi-fi equipment, including that for stereo, has had to wait and wait, while basic ear-matters of evaluation have slowly sorted themselves out in prolonged and repeated study, as described in previous issues. For how can one evaluate stereo equipment, if one isn't too sure just what stereo is and means, in fundamental listening terms?

Thus, though I'm still feeling an urge to make everything tentative, I'm now getting to the point where I dare mention equipment, and be honest. I still am not clear on many a point and, ideally, would be glad to wait a year—two years—before making up my mind. By that time most of the stuff will be obsolete, so I'm stuck with my opinions as of right now, for better or worse. My apologies to the future, just in case.

KLH Six

I've been using two sets of speakers for my stereo, one in the country and one in the city. Though I had intended to intercompare the two sets in both locations, logistic problems haven't allowed it. I'm not too sure it matters. (I feel the same way about stereo discs versus their mono equivalents—the comparison really doesn't prove much other than that both were obviously made at the same recording session and, superficially, sound very much the same!)

In the country I've had a matched pair of KLH Model Six speakers, the lower-priced full-range model in this relatively expensive line of acoustic suspension systems. I must say at once that I was sold quickly on the KLH sound and have not been unsold one little bit since, though I am aware of the virtues and limitations of the breed, in absolute terms. To give due credit to the earlier and original acoustic suspension, I tried out a single KLH Six first of all as an AB with the AR-1, and was really amazed at how very much alike their sound was in the over-all.

That merits a moment's thought. All good speakers today are claimed hi-fi, and all speaker accounts imply that this means a faithfulness to the input signal in literal terms. But as we know, speakers in general remain remarkably individualistic, both from model to model and from company-sound to company-sound. Coloration is a polite word for it; brilliance is another—presence, impact are still further terms that generally indicate what is literally a distortion of the true sound. These are not necessarily faulty sounds for the listener, since it's well known that many listeners derive legitimate pleasure from some types of speaker coloration—there is, at least for the moment, an enhanced sense of realism and presence, a better musical illusion, a more effective getting-over of the sound-sense. Tastes differ and change in these respects, as do speakers. It is obvious that we have not yet admitted the principle that ideally *all speaker sound should be alike*, i.e. non-distorted, nor its acid-test corollary, that if two speakers sound different, then one or both must be distorting the true signal.

So—I am astonished, and also pleased, to find that the over-all, in-the-large, sound of the KLH and Acoustic Research speakers is really much alike. This to me means exactly one thing: these speakers do not produce any major, large color differences. (The main tonal difference in speakers is of course in middle and high highs. Bass "coloration" is technically similar I suppose, but for the ear it is a very different matter, and in musical listening far less crucial than treble color.) Other speakers, of other brands, show among themselves—and even from model to model within one line—really major contrasts in color. Just AB a few and see.

Once this was established, I went on to find that in relatively lesser ways, the KLH and AR speakers do of course differ. The AR-1, at a higher price, a lower efficiency, gets a lower, fuller bass when you come to more specific detailed comparison. That is to be expected. I can tell the difference quite definitely, via AB listening or even after a good lapse of time. But musically, I tend to forget it quickly when my mind wanders back to the music itself.

AR-2, the lower-priced model, *vs.* KLH Six? Well—I tried them AB and am searching for words that will get over the idea that, though they are somewhat different in sound, they have similar values, not to be stated in black or white terms. The KLH Six, more expensive (it falls between AR-1 and AR-2) is slightly bigger in size and has a fuller sound; I like its ultra-smooth treble range, too; it has more "presence" in that range, and I'll not even guess as to whether that presence is in the technical sense more, or less "flat." I really don't know. KLH also has a fuller bass sound, a rounder over-all effect. But, I quickly add, the bass end for my ear seems less tight and sharp than AR-2, though not objectionably so in any way. (You see, if I say the bass booms, you'll think I'm condemning the thing. It doesn't "boom," if by that word you infer an uncomplimentary description. It has merely a pleasantly full, round bass sound and, frankly, I don't really care whether it is absolutely 100 per cent flat or not. Whatever KLH has here, it is musically satisfactory to my ear.)

The KLH pair makes a near-ideal average-home stereo set-up. I'll extend that quickly and say that any matched pair of acoustic suspension speakers makes a near-ideal set-up. Small, inconspicuous, easily moved, nicely balanced in tone, blending together beautifully, without major aberrations in color in any part of the range, these speakers, all of them, tend to maximize the effect of the music and minimize the presence of the stereo system itself. Don't use them if your interest is primarily in hi-fi outward stereo dramatics. Do use them if you want to listen to the stereo content with minimum interference from the machinery, maximum portability and convenience.

Eico

Back to Eico again. A year or so ago I talked about the interesting little Eico speaker system with the flower-pot, upwards-aimed tweeter element, spraying the high range up at the ceiling from a small square-column box like a sawed-off grandfather clock. For stereo trial, I got hold of a pair of the current production models of these, since the idea of stereo with these unusual upward-facing speakers seemed interesting.

You'll remember that (aside from the hazards of spilling beer or cocktails or even, possibly, milk into the speaker) this unit showed interesting possibilities for disbursing the apparent sound-source in a small room—where the usual horizontal tweeter system doesn't have much chance for good reflection. That was for mono sound, and the idea worked out well. Almost any location of the economically small speaker—not even a foot square at the base—seemed to produce a very good sound in a small room.

How about stereo? Well, my findings are somewhat different. First, let me say that these Eico speakers, with the flower pot arrangement still in force (the mid-highs variable but the high-highs fixed), are in present production rather on the brightly colored side. Not too much for my somewhat sensitive ear—I dislike highly colored highs, as you know. But I found that the Eicos were quick to show up any sort of distortion, scratch, hiss and the like, where the various acoustic suspension speakers with their more discreet cones tend to minimize such things, if and when.

On the other hand, when conditions were just right—and this took a long, long time—the Eico speakers did a brilliant job in the best sense of the word. Some stereo discs, I found, came through with considerably more musical impact on Eico than on my KLH country set-up. Others, however, tended to be shrill and squeaky, due to their own internal faults, somewhat touchily magnified.

(You see how hard it is to say "best." For some musical purposes the Eico pair did a better *musical* job on my stereo disc; for others, KLH came out ahead.)

I found some very interesting things about stereo placement by means of these somewhat sensitive Eicos. I had a really poor stereo set-up in the city, I found out. In the original arrangement, the Eicos squealed at me reproachfully, the sound was awful, and frankly I didn't know where to put the blame. (This was part of my early stereo agony, already described in previous issues.) It was partly Eico's fault.

But when I found a new and much better placing, the two Eicos suddenly began to sound just fine. Amazing! Nothing was changed but the speaker location—yet I now felt that the brilliance in the high end which had seemed to me insupportable in the old place was in most cases quite acceptable and in some cases actually helpful.

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to the ceiling is definitely *not* satisfactory. Not for me, anyhow. There is a very curious lack of definition, a fuzziness of spatial concept, a blurring of the sonic image—and the orchestra is oddly misplaced at best, seeming to float vaguely above you on a sort of pulsating platform. (It's more like a pulsating cloud with the instruments undulating upon it.)

The solution to the problem was very simple and it'll only cost you the temporary use of two records. Just make a 45-degree reflector on top of each Eico speaker by placing an LP record against the rear of the upward grill top, a supporting pencil at the forward edge to hold it at a slant. The instant I did this, my stereo sound jumped into place. Absolutely no doubt about it—you must have your highs straight forward and out, for good stereo. Not up.

One mild caution if you use Eicos, for stereo in an economical space. My experience so far is that the speakers tend to vary slightly in sound, by a small degree—which is getting smaller as the manufacturing techniques for the tricky flower-pot tweeter elements are standardized. My two units were hastily assembled to meet my own deadline for an upcoming demonstration last spring (and my thanks to Eico for help in an emergency); they were later re-worked with current production speaker elements and are now almost perfectly matched in sound; but I still detect a very slight color difference, and one unit seems slightly louder than the other. Easily checked—just compare your two speakers.

Stereo Reflection?

This brings me to a major side-point that must be hit right here. After a dozen years of recomending reflected speaker sound—with speakers normally turned sidewise or even straight away from the listening point—I now am entirely sure that *for stereo*, reflection is definitely a poor idea and doesn't work. My stereo speakers *always* will face straight ahead (though not converging on a point; parallel beams seem best to me in most situations).

The essence of stereo is a fusion between speakers, brought on by the sound itself. The more positive the beaming, the more positive the stereo spread between the speakers. Seems like a contradiction but it isn't. On good stereo you should not hear the speakers at all as point sources unless the recording people intend you to—as they often do, particularly in pops and jazz stereo. But, in classical, if the stereo engineer intends it, you hear your sound spread out between and behind the speakers. (IF YOUR SPEAKERS ARE IN PHASE.)

An odd side-effect of this is the frequent impression I get that my two speakers are absolutely silent, as I look at them. There is not the slightest audible evidence that the heard sound has anything to do with them at all! That's why you don't need to avoid point-source effects, by reflection. There aren't any.

(P.S. You can improve a bad stereo pickup by deliberately turning your speakers around, to widen the apparent sound source by reflection from the wall. Some ill-advised classical stereos have ultra-close-up soloists who insist on inhabiting the inside of the speaker box, on one side or the other. Reflection will spread them out at a slightly greater distance and achieve a better sonic placement—sometimes. Not important for most stereo listening, which should be strictly from the speakers' face.)

Mono Reflection, Mono Cartridge

Ah—but what of the monophonic record as played via two stereo speakers? The great plug these days is that our mono records (a) will play very nicely via stereo pickups, thus are not obsolete, and (b) that their sound is enhanced by the use of two speakers.

As to the first proposition, I am quite firm at the moment. Yes, ideally it is quite proper to play all mono records on stereo pickups. After all, the mono disc puts the stereo cartridge to work in its normal fashion, with merely a lessened vertical movement (there's some, due to pinch effect) relative to the lateral movement that is the main information carrier on *all* discs. But in a strictly practical sense, right now and for the present, I find that it's still very satisfactory to play my mono discs with one or another of the excellent mono cartridges I have hanging around. In fact, my playing system is deliberately rigged for that.

Why? Well, in the actual playing, counting in the over-all system, the mono sound is still just plain better via a top-rate standard mono cartridge than via any stereo cartridge set-up I have yet tried. So—I use my excellent standard cartridges, when and if.

The reasons, of course, are in good part outside the stereo cartridge itself. The simple fact of higher output in most mono magnetics as compared to stereo, and the use of one uncomplicated channel instead of the two tricky side-by-side channels, is enough to account for most of the difference. Yet over and above such problems, there still seems to be a residue of subtle, unexplained difference between stereo and mono cartridges, a general difference spreading over numerous brands. The "mono-mono" combination just works better, as of now.

I suspect that in a year or so this difference will have entirely disappeared and the ideal compatibility between stereo and mono cartridges, for the non-stereo disc, will have become an actuality. Indeed, the change will be largely outside of the cartridges themselves—via better connecting circuits, better shielding and ground systems, better amplifier facilities. So don't throw out your old cartridge yet—unless you're very happy with your new stereo set-up for all your old records.

As to reflected mono loudspeaker sound on a single speaker versus two-speaker mono sound, I am also somewhat surprised to find that I prefer the reflected sound of a single speaker to the new, direct two-speaker mono sound, heard when mono discs are played on stereo systems. The reason is tricky, but clear enough. The two-speaker system, far from spreading out the sound over a wider area, actually *compresses it*, into an area midway between the speakers. (That is, when the speakers are IN PHASE.)

The trouble with a single speaker pointing straight at you, as I've said for years, is that there is too often an apparent point-source that brings all the music to you from one place, though it should seem to come from a wide area. Reflection, as I've recommended it, spreads the sound-source, for a more natural and musical illusion. But with a pair of speakers the situation is radically different. There is no point-source at all at either speaker (if they are IN PHASE). Instead, the entire sound seems to come from a relatively small area right in between—and this area is *not nearly as wide*, you'll discover, as the actual spacing of the speaker units.

The centering effect, indeed, is so strong that the musical image tends to be compressed and over-narrow, a sort of spatial blob—and not only the music itself; the room-sound is also compressed. The entire

(*Continued on page 71*)

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EDITOR'S REVIEW

THE HOLE IN THE MIDDLE

One of the mysteries of the entire stereo problem is the famous "hole in the middle," which continues to plague many who have converted to this newest and most exciting of hi-fi phenomena. This is a problem which baffles us completely, because we do not believe that it should exist—except that it does, especially in some of the demonstration rooms at the various high fidelity shows, and most notably at the recent one in New York.

We are among those who have converted an existing monophonic system to stereo without junking all previous equipment. We had been using a very good corner speaker in our listening room for monophonic material, and when the urge came for conversion we simply added another high-quality speaker—one not of the same make, but of at least equal quality. Now it is admitted that these two speakers do not sound exactly alike, but either one of them alone would be judged as above average. Both have good response down to below 40 cps, and both are capable of reproducing sounds above our normal hearing range. But the two speakers do differ in over-all coloring to a certain extent, and anyone hearing both would be certain to agree.

The configuration of our listening room is not particularly unusual. The corner speaker is, naturally, in a corner. The other, not a corner model, is located on the flat of a wall—one of the walls bearing on the corner where the other speaker is located—but also at a corner. The second speaker is "aimed" directly out from the wall, and along the adjacent side of the room, which is roughly 14 by 20. Thus the centers of the two speakers are approximately 12 feet apart. We have, to be sure, heard reproduction which would be described as having a "hole in the middle," but only for so long a time as was required to reverse the phase to one speaker. This was not unusual with tapes, and we must admit that this has often caused no little confusion. At the beginning of disc reproduction, we noted the same trouble occasionally, and many other observers also reported the same troubles when switching from monophonic to stereo reproduction. However, later releases of stereo records have been consistent in phase relation—after all, the RIAA standard calls for a lateral modulation when the two signals are in phase—and our troubles have ceased.

We say unequivocally, however, that there is no excuse for the hole in the middle *if the two speakers are in phase*. But that is a very strong *if*. In the first place, some means should be provided for changing the phase of one of the two systems, and while the speaker circuit itself is the simplest place, we made a small conversion to our current test preamp—Pilot SP-215—so that it provides for a change in phase to the "B" channel power amplifier.

While a 12-foot spacing along a 14-foot wall is nearly as bad as can be imagined, we have never been conscious of the hole in the middle after we were sure the phasing was correct. This is of considerably more importance than seems to be recognized, for the phrase continually crops up in the writings of many who have learned about stereo within the past year or so. Please forgive us for harping on this subject.

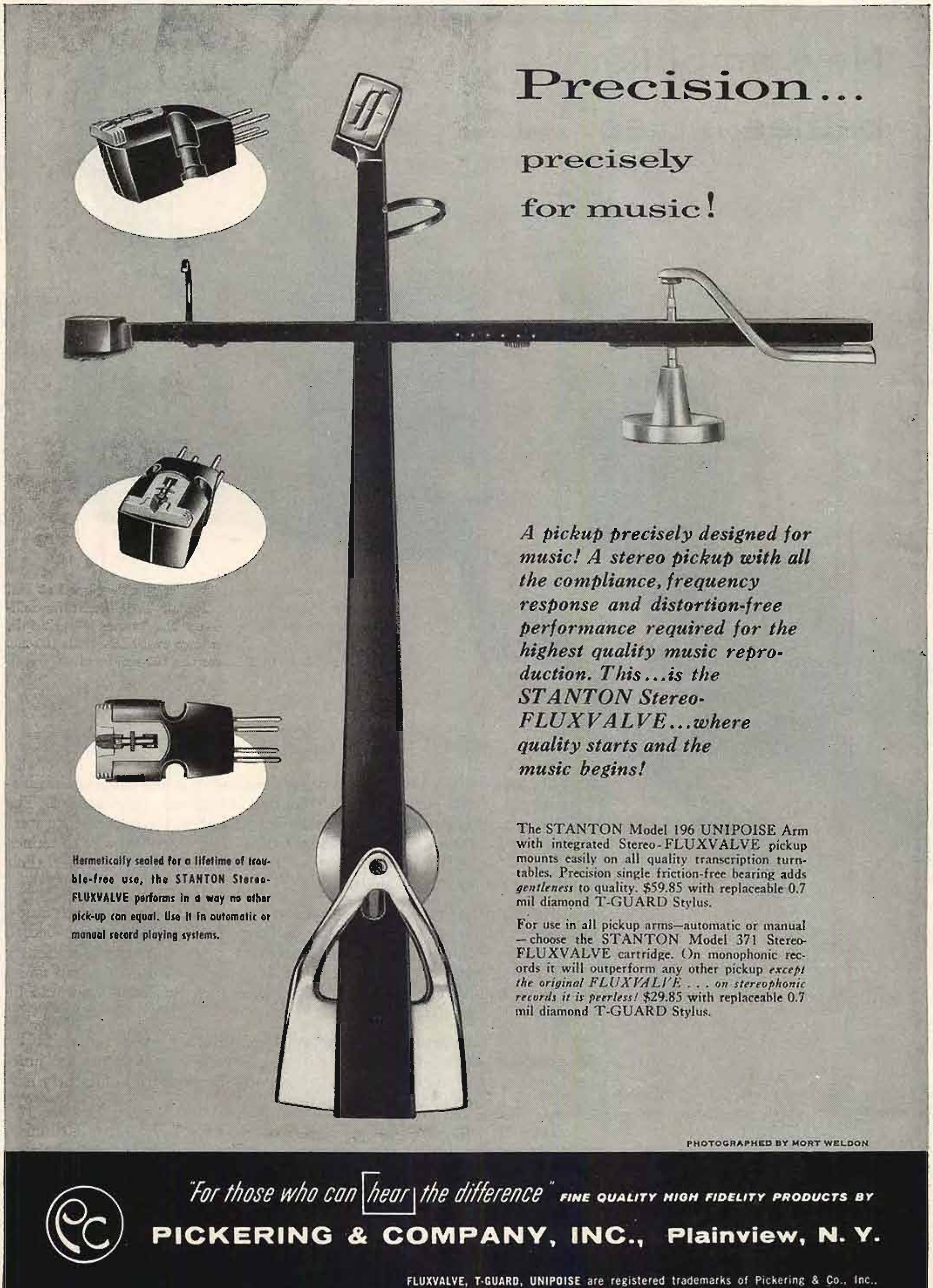
The cure? Play monophonic records through your system until you are sure that you recognize what these should sound like and where the sound source should be—right in the center, of course. If it is not, try reversing the leads to one of the speakers. You can recognize the difference by standing at the convergence point of the two speakers, and then sway slightly from left to right. If the sound appears to jump from one speaker to the other, try reversing the phase. The sound should appear to come from the space in the center when listening to a monophonic source. Once it is correct for single-channel reproduction, it should be correct for stereo, and as you move across the front of the speakers the sound transition should be smooth, and not in "ripples." Try it and see, and then forget the "hole in the middle" myth.

STEREO PICKUP HUM

In professional circles, the subject of grounding is quite important, and very little information is available on the subject—nearly everyone "cuts and tries" each time he installs a system. With single-channel hi-fi systems, the grounding problem was seldom encountered, because everything was quite straightforward.

In stereo systems, however, things are different, particularly when separate amplifier systems are used. The unavoidable leakage from power-line circuits to the chassis of one amplifier attempts to get to ground via the interconnections to the other chassis, which has its own leakage problems. Leakage is not a *fault* in an amplifier, because there is some capacitance between the windings of the power transformer and the chassis, and a minute current flows through this path. When the two amplifiers are connected together only through the shields of the two cables leading to the phonograph pickup, there is a small voltage drop across the resistance of the shields. This would be of no trouble in a speaker circuit, but in a circuit where the highest voltage is likely to be of the order of one-hundredth of a volt, it can cause trouble. Try connecting the two amplifiers together through a low-resistance lead, such as an automobile battery cable. If hum still persists, disconnect the shield of *one* of the pickup leads at the turntable or changer end, with only the "hot" lead connecting to the input of *one* of the amplifiers.

These suggestions should help in most instances. If your hum is constant no matter where the pickup is with respect to the turntable, these methods are usually effective. If moving the pickup over the turntable varies the hum, that is something else again—inductive pick-up into the pickup. You will need some effective magnetic shielding to cure this, and your own experimenting should lead you to the right answer.



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Phasing in Stereophonic Recording

WILLIAM S. BACHMAN*

A clarification of the confusion which seems to exist with respect to the phasing of stereo signals so as to reproduce correctly in the listening room. Of more importance than commonly recognized, phasing is easily corrected, and standardization will soon completely eliminate the problem.

The problem of phasing the two channels of information in stereophonic recording and reproduction is the same, regardless of the use of tape or disc as the recording medium. Because disc records afford rapid comparison between monophonic and stereophonic program, errors in phasing are much more readily observed, even though their effects would be of equal importance in tape reproducing systems.

One of the principal factors leading to the choice of the 45/45 system as the R.I.A.A. Standard was the fact that in-phase information on the two channels could be made to appear as lateral modulation, making the system completely compatible with existing lateral monaural recordings. When two microphones are used with close spacing as they are in the X-Y system, *Fig.* 1, the dominant acoustical information is in-phase, particularly at the low frequencies. According to the R.I.A.A. Standards, this in-phase information should appear on the record as lateral modulation. In the M-S system of recording, *Fig.* 2, using vertical-lateral channels as opposed to 45/45, the output of the "M" microphone appears as the lateral channel. While the vertical-lateral stereophonic disc recording system is not standard, the M-S microphone technique can be adapted to a 45/45 recording system by adding the microphone outputs vectorially. This is completely rigorous and results in the "M" microphone output appearing as lateral modulation. In this way, the compatibility of the R.I.A.A. Standard with the two systems of stereo recording is clearly shown. As the microphones become more widely separated, the phase of the acoustic signals reaching them becomes more random. While it is desirable to maintain the phasing even so, it is more difficult to ascertain by listening whether or not it is correct.

* Director of Research and Engineering, Columbia Records, a division of CBS Inc., 799 7th Ave., New York 19, N. Y.

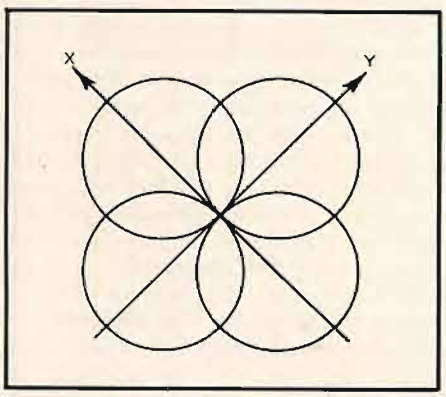

Fig. 1. "X-Y" type of studio pick-up. Two figure-eight pattern microphones, such as ribbons, are used 90 deg. apart, each inclined 45 deg. to the sound source. The two outputs become the right and left modulations of the 45/45 stereo groove.

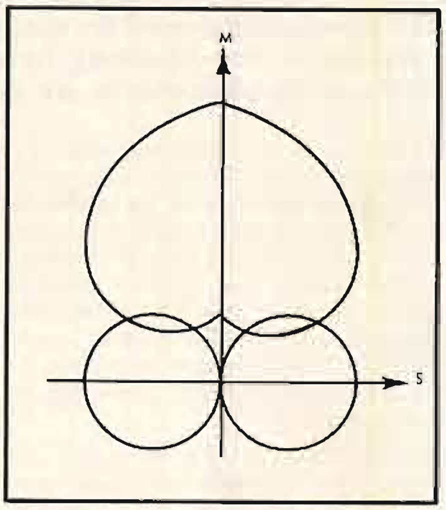

Fig. 2. "M-S" type of studio pick-up. As applied to vertical-lateral stereo disc recording, the M microphone feeds to the lateral channel, and the S microphone feeds the vertical channel. For 45/45 stereo disc recording, the two microphone outputs may be combined vectorially as M + S and M − S. In either case, the M microphone output appears as lateral modulation on the disc.

The two channels of a stereo recording system can provide a third channel by means of a phantom circuit. This is done by feeding the third channel information equally to the right and left channels. This is analogous to the well known technique in telephone systems in which three conversations are carried on two pairs of wires. If this phantom channel is to be effective, the phasing of the system must be rigorously maintained. Since this phantom information is applied equally to the two channels in-phase, it will, therefore, appear on the disc as lateral modulation.

Equal Gains

In the reproduction of stereophonic signals, it is necessary that in addition to maintaining the phase, the gain of the two channels be made equal. If the two channels are equal in gain and frequency response and are displayed on identical loudspeakers, a sound image is formed half way between the loudspeakers for in-phase signals. In a disc stereophonic reproducing system, this is readily observed by the use of a monophonic record as a signal source. Such a record has only lateral information, and if the phasing of the reproducing system including the loudspeakers is correct, the gain of the two channels can be adjusted to place this image very accurately between the loudspeakers. If the loudspeakers are dissimilar, the image will not be as sharply defined, tending to move as the frequency of the program material varies. Reversing the phase of one channel destroys the image completely. While this is not an altogether unpleasant effect, it will be accompanied by a loss of bass if the speakers are not very widely separated.

It is apparent from observations of the stereophonic discs now on the market that some depart from the RIAA Standard which states that in-phase information should appear as lateral modulation. This, in many cases, is probably due to accidental phase reversals in connecting tape machines for editing and re-recording, or due to the phasing of the microphones themselves. Rotating a ribbon microphone 180 deg. reverses the phase. It may also be due to confusion arising from the cutter schematic diagram indicating the two recording motors as vec-

(*Continued on page 74*)

Improvement in "Air Suspension" Speaker Enclosures with Tube Venting

PHILIP B. WILLIAMS* and JAMES F. NOVAK**

According to these authors, the vented enclosure can furnish more and cleaner output, theoretically, in the octave band around the speaker resonant frequency, with performance above this region then becoming equal to that of the closed box. Design factors and comparative performance are shown for both systems.

If the title of this paper sounds controversial, its purpose has been achieved. In these days of "all-new" enclosures, "revolutionary concepts" and "totally new principles of acoustics" a clear-cut comparison of virtues is too seldom a part of the sales story or of an article describing the product. Nor do we see acknowledgment of the usual close kinship or identity with enclosure systems described and analyzed long ago in such classics on acoustics as handbooks by Olson, Beranek and others. There has been no basically new type of enclosure in this decade, but much worthwhile effort has been devoted to refinement and improvements of existing basic types. There are few basic types of enclosures for the dynamic speaker. These are, as shown in *Fig.* 1:

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1. *Infinite Baffle:* The dictionary still defines infinite as quite large. The growing popular usage of this term for the small box should be discouraged.

A practical measure of the size of the box type infinite baffle is that volume beyond which further increase makes no appreciable difference in compliance of the air behind the speaker. A true infinite baffle does not raise the resonant frequency of the speaker over that measured in free air. It actually lowers the resonant frequency slightly, due to additional air loading as the back wave and front wave are effectively separated. For most 15-in. speakers, for instance, a 14-cu. ft. enclosed box is infinite in a practical sense.

A flat-surface infinite baffle is one that effectively separates the speaker front wave and back wave above a criterion frequency, to avoid output cancellation. The flat baffle acts as an infinite baffle when it makes the external air-path length between front and back of the cone a minimum of one half wavelength. An open-back cabinet is a folded flat baffle, at least until the depth exceeds ⅛ wavelength.

2. *Closed Box:*[1] Here is implied a box small enough to have appreciable effect on the resonant frequency of the speaker because of the stiffness of the air behind the speaker. Speaker resonant frequency is higher than in free air. Intensive and productive effort in the last few years has been devoted to optimizing the elements of reproducers in the very small classification.[2] One important result of such developments has been an effective change in the thinking of audio people about loudspeaker efficiency. Some years

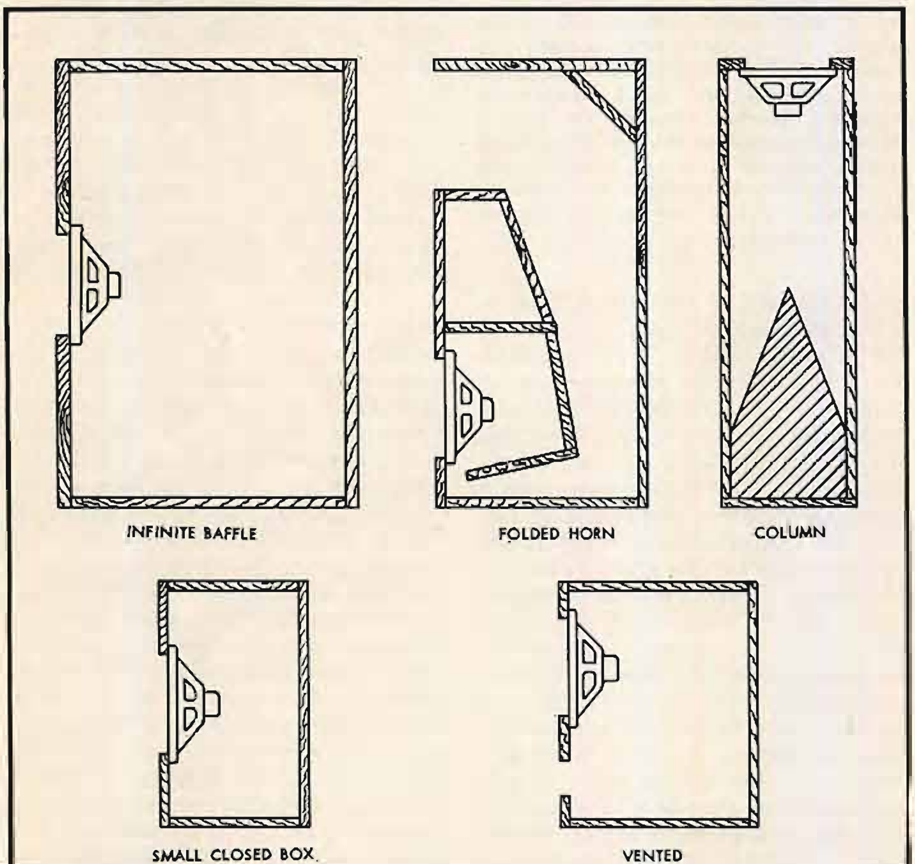


Fig. 1. Basic types of speaker enclosures.

[1] D. A. Dobson, *"Closed-Box Loudspeaker Enclosures."* Master's thesis Electrical Engineering Department, Massachusetts Institute of Technology, (See *"Acoustics,"* L. L. Beranek, p. 221).

[2] E. M. Villchur, "Commercial acoustic suspension speaker." Audio, July, 1955.

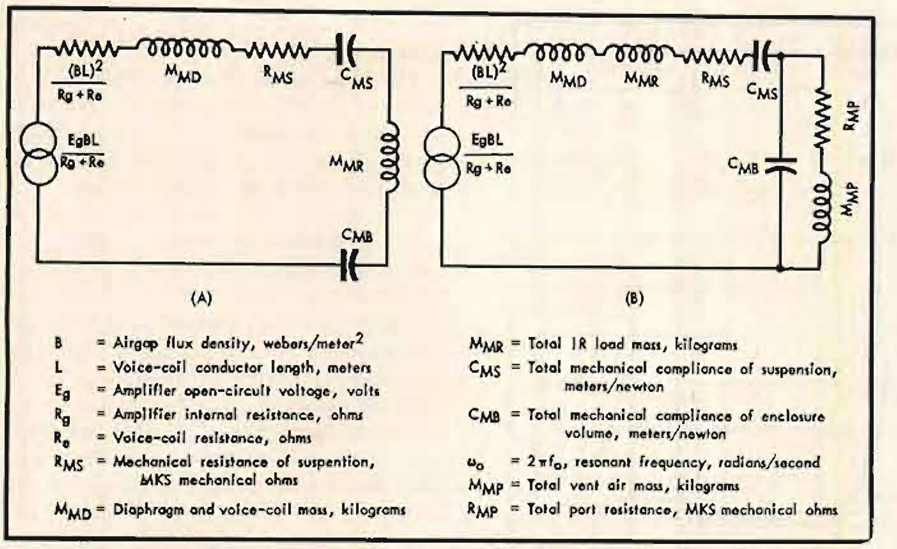


Fig. 2. Mechanical equivalent circuits of direct radiator speakers in: (A), closed box, (B), in vented enclosure.

ago, equipment salesmen would have been horrified at the idea of selling very low efficiency loudspeakers which would need special consideration in amplifier design or rating. Now the idea of adequate bass in a small space is attractive enough as a sales point to offset the special complications of higher power or low damping factor, or other complications which increase the amplifier cost. A fact of life must be pointed out here, however. A large box always allows *more* and cleaner bass than does a small box.

3. *Horn:*[3] True horn loading increases the radiation resistance loading and output of the speaker over a relatively wide frequency range. There is no other enclosure which can deliver so much output at such low distortion, and over such a wide frequency range. Many enclosures labelled as horns are deficient in flare path length, mouth area, or other characteristics, so that they operate in some manner other than as a classical horn. Their deficiencies generally are the byproduct of economy in size. An effective horn, at least at this stage of the acoustic art, cannot be small if it is to reproduce bass.

4. *Column:* This can have either an open or a closed end opposite the speaker. The closed column might be considered a special case of the closed box in which one dimension becomes large enough to decrease the reactance presented to the speaker. The labyrinth is an open folded column using radiation at the open end to reinforce output from the speaker end. The column enclosure is usually resonant to a significant degree.

5. *Vented Box:* This uses the back wave of the speaker over a range of about an octave or one and a half octaves to reinforce output from the front. It consists of a closed box with a hole, or series of holes, or a duct leading to the outside, of such reactance characteristic

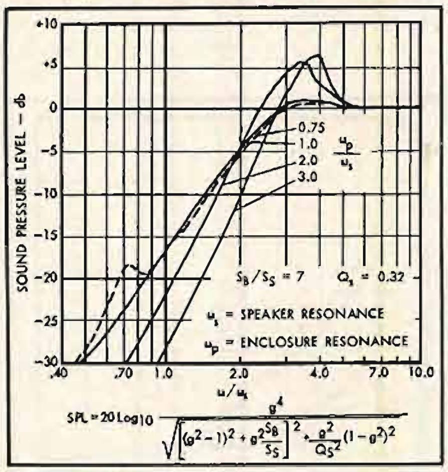


Fig. 3. Relative sound pressure level response of a direct radiator in a vented enclosure as a function of enclosure tuning.

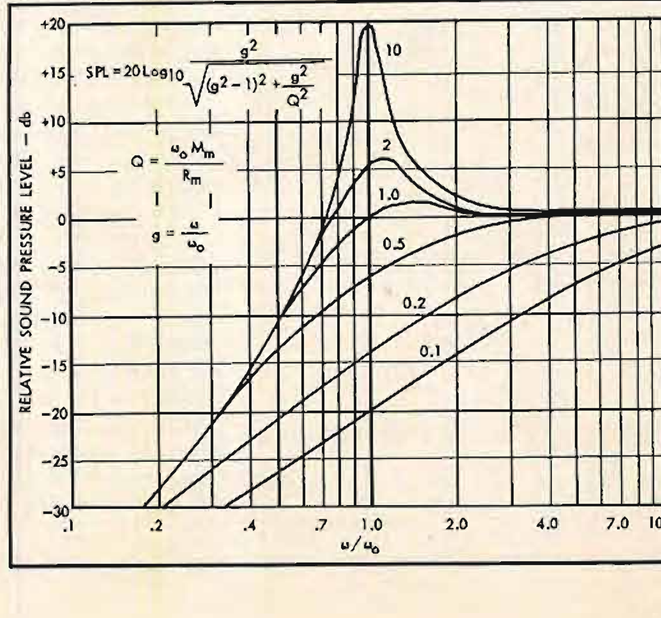
as to tune the enclosure as a Helmholtz resonator. The frequency of tuning has a planned relationship to the resonant frequency of the speaker, most frequently being equal to the speaker resonant frequency. Below the effective reinforcement range the enclosure acts as a very small flat baffle, with cancellation of frontal output by the out-of-phase rear wave. Above that range, the vented enclosure acts like a closed box, as though the port or duct were closed. Poorly designed or improperly used box-and-speaker systems have created some prejudice against venting. The appellation "boom box" must have arisen from improperly tuned combinations creating pronounced response peaks above speaker resonant frequency, as in *Fig.* 3. Of great antiquity, this widely used principle is especially devoid of glamor because it has been around so long. Often when it is used in commercial enclosures a new look is given by adding certain appendages called distributed port, friction loading, or other names describing some adjustment or modification of the elements.

Other enclosures are variations or combinations of these basic types, perhaps with a restricted outlet or some other feature to vary performance in some manner.

The old truism of "not getting something for nothing" applies nowhere more than to the mechanism of bass reproduction. Sizeable clean output at low frequencies in a well-balanced system *with good efficiency* requires that the enclosure be relatively large. The small system can incorporate all these features except one. As efficiency is the most justifiable sacrifice, we will assume a reasonable loss in efficiency for the systems to be described.

It has been apparent to most industry people that 10-watt amplifiers have sufficient peak capacity for most installations, where conventional speaker systems are used. VU meter indications of one watt or so are about as high as reached in normal listening. Something like 50 milliwatts average power pro-

Fig. 4. Sound pressure level response of direct radiator speaker; infinite baffle or closed box assumed.



[3] Technical Monograph No. 5, *"Horn-Type Loudspeakers,"* Jensen Manufacturing Company.

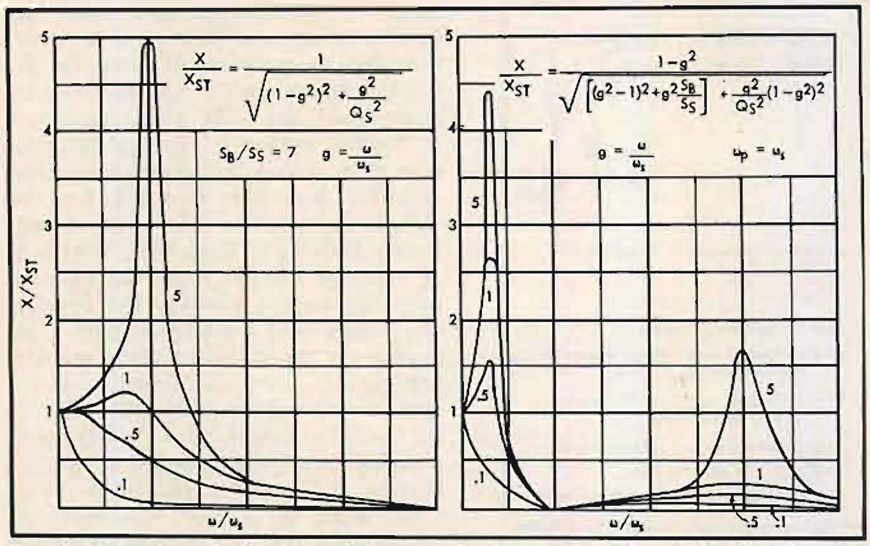


Fig. 5. Excursion of direct radiator speaker as a function of Q. (A) closed box, infinite baffle, or free air operation assumed, and (B), in vented enclosure.

duces a pleasing level of sound in the average listening room.

So we should be able to justify a loss of four to six decibels of efficiency in the speaker system before noticing distortion on peaks from a good 10-watt amplifier.

Ten decibels loss would be going too far, for a 10-watt amplifier, as there would then be no reserve to handle peaks. So we will, in this discussion, concentrate on adjustments and performance of small speaker systems requiring no more than 10 watts of power, but having bass response on a par with middle- and high-frequency response.

The Closed Box

The power radiated at low frequencies and the shape of the response curve are determined by the Q of the complete loudspeaker system, which includes the speaker, enclosure, and amplifier. (A) in *Fig.* 2 shows the low-frequency equivalent mechanical circuit of the complete closed-box system.

The resonance of the system is

$$\omega_o = \sqrt{\frac{C_{MS} + C_{MB}}{(M_{MD} + M_{MR})(C_{MS} \cdot C_{MB})}} \quad (1)$$

Mechanical Q is defined exactly like its electrical counterpart.

$$Q = \frac{\omega_o (M_{MD} + M_{MR})}{\dfrac{(BL)^2}{R_g + R_e} + R_{MS}} \quad (2)$$

Relative radiated sound pressure expressed in decibels is

$$SPL = 20 \log_{10} \left| \frac{g^2}{\sqrt{(g^2 - 1)^2 + \dfrac{g^2}{Q^2}}} \right| \quad (3)$$

where $g = \frac{\omega}{\omega_o}$ = forced frequency ratio.

From this expression are plotted several values of Q in *Fig.* 4.

It would appear from *Fig.* 4 that a Q of 0.5 corresponding to critical damping (and best transient response) does not give the flattest output down to resonance. Actually these curves apply strictly to the case where the speaker operates into a 180-deg. solid angle.

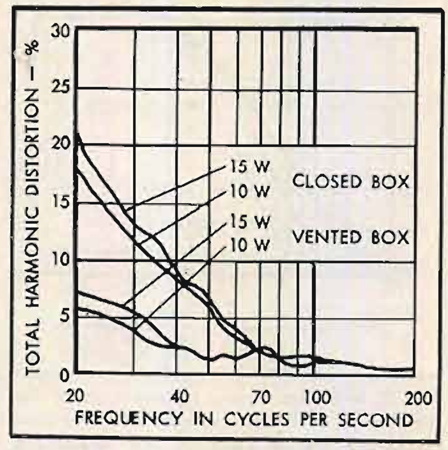


Fig. 6. Distortion characteristics of closed box vs. vented enclosure for inputs of 10 and 15 watts.

Reducing the solid angle to 90-deg. (corner operation) can improve the low-frequency performance so that output could be almost flat down to resonance.[4]

All high-efficiency speakers are overdamped when operated in infinite baffles or closed boxes with amplifiers having damping factors of 10 or greater. There is really no point in operating a speaker in an overdamped condition (Q smaller than 0.5) because transient response does not improve. One merely loses low-frequency output. Inspection of Eq. (*2*) shows that the factor exerting the greatest influence on Q (and low-frequency output) is the BL product. This is the product of magnetic field strength and voice-coil conductor length. The next most important factor is R_g, which is related to damping factor by the expression $DF = Z_s/R_g$ where Z_s is speaker impedance.

Moving-system mass is listed last in order of importance, since the range of allowable variation is comparatively small, due to manufacturing limitations, cost, and response considerations.

The closed box enclosure leaves us only one method by which to obtain flat response down to resonance, the varying of system Q. Equation (*4*) shows that the most practical method is varying the BL product. Decreasing the BL product increases efficiency at resonance but it also decreases efficiency at medium frequencies, per Eq. (*5*). The moving-system mass is usually larger than normal because of longer winding length, and this also decreases efficiency. The crux of the matter is that in order to maintain flat response, the *over-all* efficiency will have to be low.

The power-available efficiency of a loudspeaker in a closed box varies in the following manner with respect to the *BL* product and R_g:

At Resonance

$$PAE \approx \frac{R_g}{(BL)^2} \approx \frac{1}{(BL)^2\, DF} \quad (4)$$

At Medium Frequencies

$$PAE \approx \frac{(BL)^2 R_g}{(R_g + R_e)^2\, (M_{MD} + M_{MR})^2} \quad (5)$$

The Peak Amplitude of the Cone Movement in Meters

$$X = \frac{BLe_g\, C_{MS} \sqrt{2}}{(R_g + R_e) \sqrt{(g^2 - 1)^2 + \dfrac{g^2}{Q^2}}} \quad (6)$$

A plot of Eq. (*6*), normalized, i.e.,

$$\frac{X}{\dfrac{F}{S}} = \frac{X}{X_{static}} \quad (7)$$

where

$$F = \frac{BLe_g}{R_g + R_e} \text{ and } S = \frac{1}{C_{MS}}$$

is shown in (A) of *Fig.* 5.

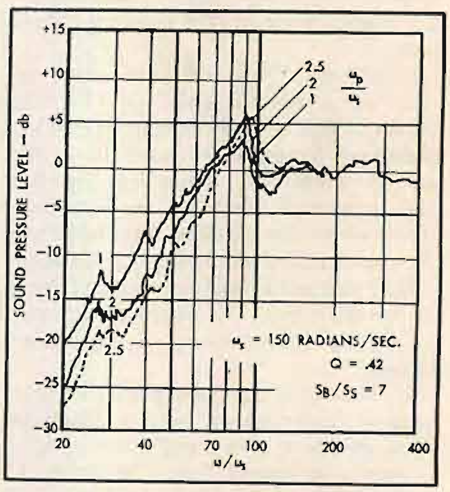


Fig. 7. Experimental data showing relative sound pressure level response of direct radiator in vented enclosure as a function of enclosure tuning.

[4] Harry F. Olson, *"Acoustical Engineering,"* 1957, pp. 31–32.

Distortion

Distortion is generated in two ways:

1) Non-linearities in the suspension
2) Non-linearities in the magnetic field

A popular misconception about item 2) should first be examined. It is generally believed that making a voice coil long and thereby maintaining a constant number of turns in the gap automatically ensures against distortion due to non-linear driving force. This is not so because the magnetic flux does not cease to exist outside of the air gap. The fringing flux must be taken into consideration because the average number of lines of flux cut in both directions by *all* the voice-coil turns determines the amount of distortion. This means that the flux distribution over the entire winding length must be symmetrical.

The force/displacement curve of a typical loudspeaker cone-suspension sysstem is linear only for small amplitudes. Merely making the suspension highly compliant will not ensure linearity over large excursions. Consider a perfectly flat suspension, i.e., one with no rolls. This suspension has the highest compliance of any, but is also the most non-linear even for fairly small amplitudes. Obviously, the suspension must change its length in a linear fashion if it is to allow large cone amplitudes. The proper choice of roll design, cloth, and resin will give a highly compliant suspension that behaves like an accordion bellows and yet permits excellent lateral stability.

The result of a non-linear suspension system is production of odd-order harmonics with the third harmonic being predominant.[5] Because the amplitude of a direct-radiator speaker is inversely proportional to the square of the frequency below the region of ultimate radiation resistance, greatest distortion will occur at the low frequencies.[6]

It was shown earlier that in order to maintain flat response down to resonance, the speaker Q would have to be near 1.

(A) in Fig. 5 also shows that the excursion gets quite high in the region of resonance and below.

Since even the best suspension will have some degree of non-linearity, an ideal enclosure would be one which kept the cone excursion at about the same magnitude as at medium frequencies and would still give at least as good response as the closed box.

The Vented Enclosure

A vent in the closed box adds a second degree of freedom to the system, causing a redistribution or repartition of the resonant frequency and damping of the speaker. The original speaker resonance is replaced by two other resonances, one near the closed box resonance and one substantially below the speaker resonance. The damping at these two new resonances is greater than for the closed-box case.

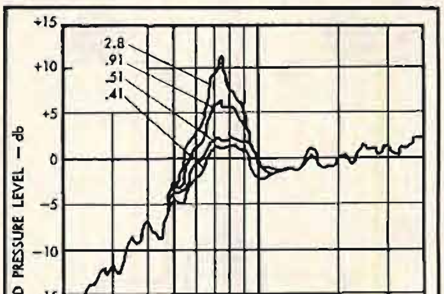


Fig. 9. Experimental data showing relative sound pressure level response of direct radiator in vented enclosure as a function of speaker Q.

Fig. 10. High-compliance, long-travel Flexair woofer. Resonant frequency 22 cps for 12-in. size, 18 cps for 15-in.

The vent in the enclosure acts as an acoustical mass which resonates with the compliance of the enclosure at some particular frequency. An investigation of the equations of motion for this enclosure shows that this particular resonance should be equal to the speaker resonance of maximum output is to be obtained down to speaker resonant frequency. This is shown theoretically in *Fig.* 3, and experimentally in *Fig.* 7.

An exact analysis of the equivalent circuit is quite complicated because dissipation exists in both speaker and port meshes and because mutual coupling exists between the speaker and vent. This mutual coupling, which is a function of the size and spacing of the two radiating surfaces, alters the radiation impedance. From a practical point of view, if we recognize that in most instances the port Q will be 10 to 20 times greater than the speaker Q and that the presence of mutual coupling will generally enhance the output, we can ignore the effects of both. This greatly simplifies the equations of motion and allows the calculation of theoretical response with only small errors.

(B) of *Fig.* 2 shows the low-frequency equivalent mechanical circuit of the complete vented enclosure system. The approximate resonant frequencies of the system are:

$$\omega = \frac{\omega_S}{2}\sqrt{\left[2+\frac{S_B}{S_S}\right] \pm \sqrt{\frac{S_B}{S_S}\left(\frac{S_B}{S_S}+2\right)}} \quad (8)$$

where $S_B = \frac{1}{C_{MB}}$

$S_S = \frac{1}{C_{MS}}$

(*Continued on page 75*)

[5,6] Harry F. Olson, *"Acoustical Engineering,"* 1957, p. 186.

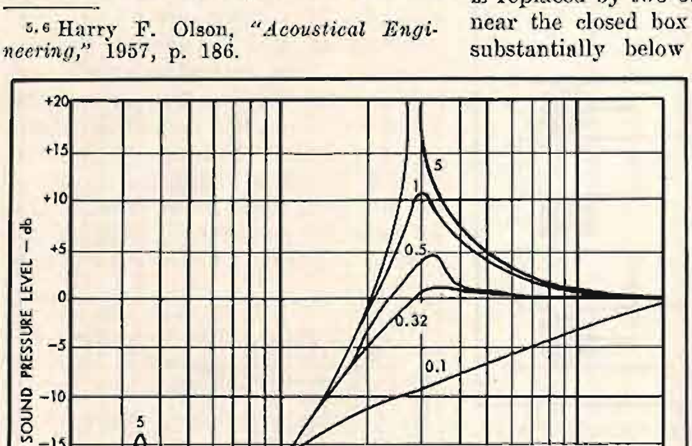


Fig. 8. Relative sound pressure level response of direct radiator in vented enclosure as a function of speaker Q.

For Stereo, The Bi-Ortho Output Circuit

C. NICHOLAS PRYOR*

A method of obtaining two signal channels in a single set of output tubes by use of their orthogonal connections, and an analysis of the circuit application to stereo reproduction.

Several years ago an old trick of the broadcast-line engineers, the "Simplex," was put into use in a push-pull audio amplifier to provide an additional utility amplifier channel with very little modification to the existing amplifier.[1] The principle of this trick is shown in *Fig.* 1. At (A) is shown the broadcast line approach; the center-taps of a balanced line are returned to ground through transformer windings. The balanced connections to the line still work as before, one side swinging positive as the other side swings negative; but an additional channel of information is allowed as the simplex input drives both sides of the line together. It is easy to see that the two signals are distinguished and separated by the transformers at the output. The circuit of (B) is the same with the broadcast lines replaced by vacuum tubes, and it can now be seen that the push-pull tubes in the final stages of most modern amplifiers are capable of handling the extra signal.

The broadcasters did not stop with the simplex; when there were two balanced lines they made a third with the "phantom", shown at (C) in *Fig.* 1. Its operation is similar to that of the simplex in that the two wires in each line are in push-pull for one signal and parallel for another. Of course the center of the phantom could be simplexed for a fourth channel, etc. In general any number (n) of wires will provide ($n-1$) separate balanced channels and one single-ended channel. Any of these techniques may be applied to vacuum-tube amplifier systems simply by inserting the amplifier stages in the lines. It is usually better to mix the amplifier input signals by resistive networks or grid-cathode mixing rather than using the transformers suggested by the broadcast line analogy; transformers tend to be heavy, expensive, and often not too good.

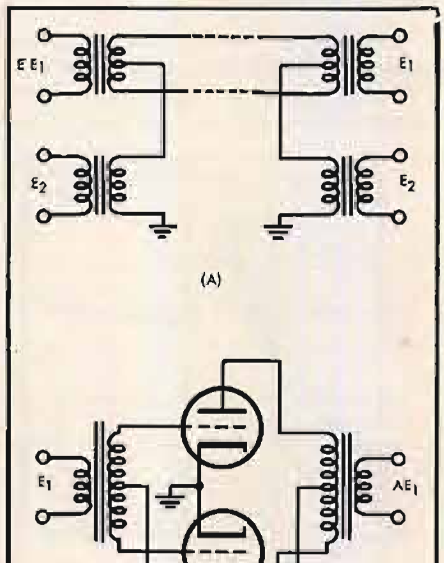


Fig. 1. Basic principles of simplexing. (A) shows circuit used in communications to transmit two signals over two wires. (B) is similar circuit arranged to work with vacuum tubes. (C) shows expansion of simplexing to transmit three signals over four wires.

Aside from the fact that the simplex can be added easily to existing amplifiers, it has certain other advantages that are worth investigating here. In particular the two-signal case provides a very interesting power balance. If we assume sinusoidal signals of different frequencies on both of the channels, the condition for staying below overload is that the sum of the voltages of the sinusoids be less than some maximum value. Since power varies as the square of the voltage, we may write the maximum available power in channel 2 as a function of the power delivered in channel 1 thus:

$$P_2 = V_2^2 = (V_{max} - V_1)^2 = V_{max}^2 + V_1^2 - 2\,V_{max}\,V_1 = P_{max} + P_1 - 2\sqrt{P_{max}\,P_1}$$

This relationship is plotted in *Fig.* 2 where the lower curve is P_2 and the horizontal axis is P_1. The middle curve shows the time average sum of the powers or the total output power of the amplifier. The powers may be added directly only when the assumption is made that the frequencies are different. When there is any correlation between the signals in the two channels, a cross-product term appears in the total power; and in the case of perfect correlation (e.g. sinusoids of identical frequency and phase), the total power is P_{max} regardless of the distribution between channels. Thus, depending on the correlation and the distribution, the total available power lies somewhere within the shaded area. (Incidentally, the same is true of any single-channel amplifier. If two equal sinusoids of different frequencies are put into an amplifier, the maximum power at the output will only

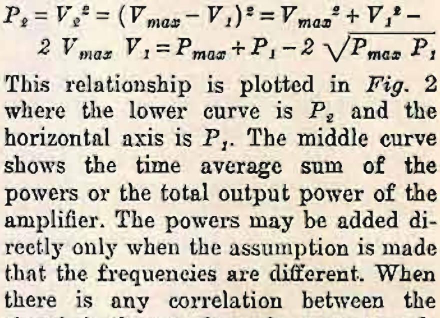


Fig. 2. Power distribution in dual-channel amplifier.

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* *4324 Clagett Road, Hyattsville, Maryland.*

[1] R. S. Houston, "Simplexing a standard amplifier for dual-channel operation." Audio, November, 1954, p. 36.

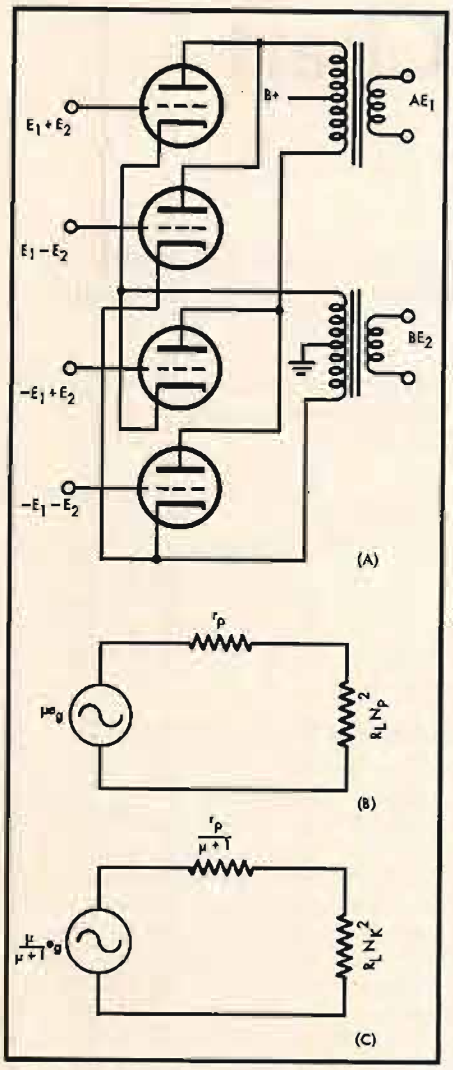


Fig. 3. Simplified circuit of the output stage is shown at (A), while (B) is the mid-frequency equivalent for the plate circuit and (C) is the equivalent for the cathode circuit.

be half the rated single-frequency power of the amplifier.)

The noteworthy feature of this power balance becomes apparent when we consider that the usual operating point of hi-fidelity amplifiers is at least 15 to 20 db below the maximum power. Consider that each channel might be operating at −15 db referred to maximum power, so that the total power output is −12 db. It may now be seen from *Fig.* 2 that *either* channel may now handle a peak at −1.5 db, a safety factor of 13.5 db. Herein lies the advantage of the simplex; if the two channels are handling uncorrelated signals, each appears to have nearly the total amplifier power in reserve. There are presently two major hi-fi applications where a pair of amplifiers is needed to handle relatively uncorrelated signals: playback systems with low-level crossover and stereo systems. It was found that the simple simplex could be applied to the former situation[2], but a push-pull arrangement was desired for both channels of the stereo system. It is for this application that the following circuit was developed.

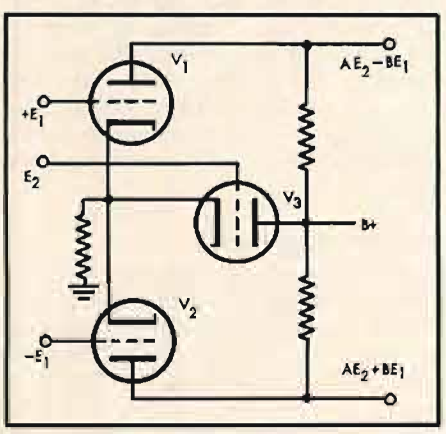


Fig. 4. Suggested grid-cathode mixer stage for feeding the Bi-Ortho output circuit.

Practical Circuit

The simplified circuit of the output stage is shown at (A) in *Fig.* 3 and is for push-pull parallel operation of the finals. It is not directly comparable to any of the broadcast connections, but may be easily derived from the phantom circuit by moving the phantom transformer to the cathode circuit and paralleling the remaining plate transformers. Since the plate and cathode circuits are independent, we may analyze each of them independently, writing an equivalent circuit for each. In *Fig.* 3, (B) is the mid-frequency equivalent circuit for the plate circuit, and (C) is the equivalent for the cathode circuit. The grid-to-grid input voltage is e_g; R_p and μ are the plate resistance and amplification factor for each tube; n_p and n_k are the turns ratio of plate and cathode transformers respectively. From these equivalent circuits we can derive the power sensitivity[3] and output impedance of each circuit. For the plate circuit:

$$PS = \frac{R_L\ n_p^2\ \mu^2}{(r_p + R_L\ n_p^2)^2}\ and\ Z_o = \frac{r_p}{n_p^2}$$

and for the cathode circuit:

$$PS = \frac{R_L\, n_k^2\, \mu^2}{[r_p + R_L\ n_k^2\ (\mu + 1)]^2}$$

$$and\ Z_o = \frac{r_p}{n_k^2\ (\mu + 1)}$$

To have as much similarity as possible between the two channels it would be desirable to equate the output impedances, thus making $n_p = n_k\sqrt{\mu + 1}$. Inserting this in the equations for power sensitivity, we note that for equal output impedances the power sensitivity of the plate channels is $\mu + 1$ times larger than that of the cathode channel. This means that the voltage drive required for the cathode channel input will be $\sqrt{\mu + 1}$ times as much as that required for the plate channel. From this it would appear good design to use a low-mu triode such as the 6AS7 or the smaller 6BX7. Each comes with two sections in one envelope; μ and R_p for the 6AS7 are 2 and 280 ohms, and for the 6BX7 they are 10 and 1300 ohms. The optimum loading for a triode is between two and three times its plate resistance; a good value for the 6BX7 is 4000 ohms plate-to-plate. This gives a turns ratio to the 8-ohm tap of about *24.2*. The cathode

(*Continued on page 81*)

[2] C. Nicholas Pryor, "Simplexing for low-level crossover." *Radio & TV News* (to be published).

[3] Power sensitivity is here defined as the power output in watts divided by the square of the grid-to-grid input voltage.

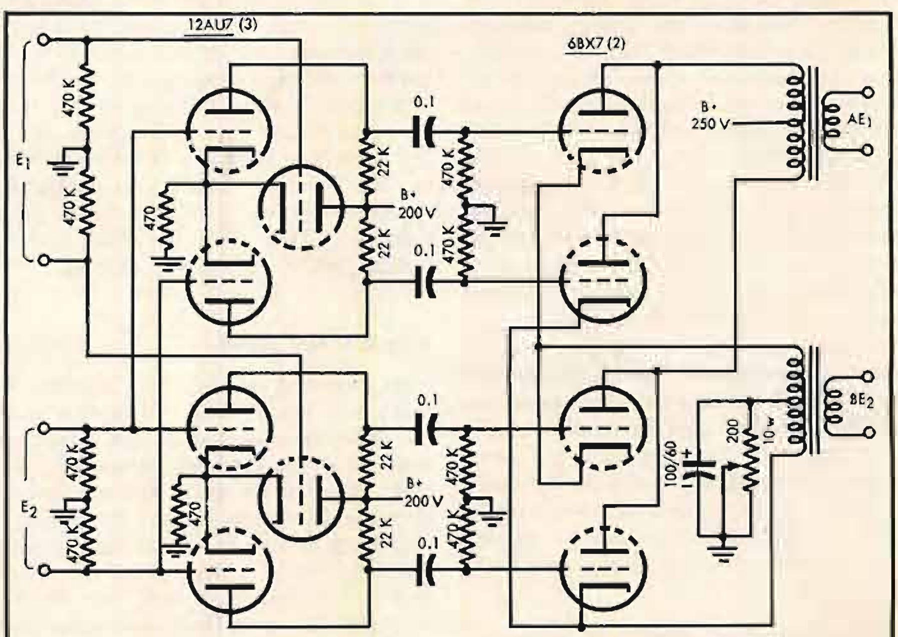


Fig. 5. Complete Bi-Ortho output and driver stage schematic.

Damping Factor Chart

PHIL PHILLIPS*

Using this chart, two resistors, and an indicating meter calibrated in db, anyone can readily determine the damping factor and internal impedance of his amplifier in a matter of minutes.

DAMPING FACTOR is a function which appears to be well understood—only by engineers, only partially by hobbyist-experimenters, and practically not at all by audio equipment salesmen. There is no reason why this should be so; damping factor is in itself a quite simple concept, and the spectacular tricks that it appears to play are only illusory. This chart, in addition to its practical utility, presents this function pictorially in a fashion calculated to erect in the mind of the average (if slightly confused) reader a clear picture of just what is going on.

The chart could be laid out more compactly by displaying the curves back-to-back instead of end-to-end, so that they both use the same DF scale—one curve sitting on top of "+ db" and "– db," the corresponding sign being applied to the DF reading. The thing looks like a badly lopsided horn sitting on its mouth, the extension of the throat being in the direction of infinity. Such a layout would permit a considerably expanded scale, or the same scale in less space; however it would present its information in a manner that could easily be misleading.

With the chosen configuration, however—one curve running from the low positive well into the middle negative region, uncomplicated by such distractions as change of scale, folding, or inverting—correct interpretation is automatic. The great positive and negative infinity mystery, for instance, is solved at a glance, and it becomes evident that infinity is a real value after all—as, indeed, it would have to be; otherwise, there could be no negative region. It is also clear that minus one is damping factor's *real* infinity; that point at which zero change in load produces infinite change in level; and the point conventionally called infinity—that point at which infinite change in load produces zero change in level, which is the same thing as saying that the circuit resistance is neither positive nor negative—is really *unity*, as indeed, these values are designated in the expression "damping ratio."[1]

Method of Use

In this method, we measure the output of the amplifier under two conditions of load, R_1 and R_2. R_1 should be equal to, or close to, the nominal output impedance of the amplifier. R_2 could be either greater or less than R_1, one choice being as good as the other, provided the amplifier is stable under all conditions of load. However, many (if not most) amplifiers are not. "Stability," as used in these notes, is intended in its general sense, applying to the degree of uniformity of all characteristics of the amplifier with changing load. Since it is popularly supposed that the normal departure of a loudspeaker from nominal impedance is in an upward direction, it seems reasonable to suppose that the designer of the amplifier expended most of his efforts toward achieving stability in this region, i.e., from nominal output impedance on up. It seems logical, therefore, to make R_2 greater than R_1, thus keeping the loading on the amplifier in the region wherein maximum stability is most probable.

Another reason for this choice is that this way the meter deflection is always upward for positive DF and downward for negative DF. The experimenter will just naturally associate + db with +DF and vice versa; besides, he may have made previous use of the method which involves disconnecting the load entirely (which is the same thing as saying that $R_2 = R_1 \times \infty$), or may still wish to use it for measuring very low damping factors or for locating infinity precisely. It would be confusing if one system used + db for – DF, while the other system had it just the other way around.

Why R_2 is Not Infinity

The method wherein one reading is taken with normal load and another with the load disconnected is not practical here. It is the inherent nature of very high negative damping factors that a very small change in load produces a very large change in level in the opposite direction. It is evident from the chart that a 2:1 change in load, represented by Scale A, is entirely too large for measuring the very high negative region. Even a 1.1:1 change, represented by Scale B, only takes us out to –1.09 (for a 6-db deflection; still only –1.03 for 12 db).

It is evident that nothing but a 1:1 change in load could ever measure Minus One Itself.[2]

The "total disconnect" method could have been used in the region wherein it would produce practical deflections, but was not because of the scaling and stability problems it would have introduced.

Choice of $R^2 = R_1 \times 2$ for Scale A

The ratio of 2:1 for Scale A is a compromise between conflicting interests. Scale A would, of course, extend further into the negative region with a lesser ratio, but then the db deflection in the positive region would be much smaller, with consequent extra reading difficulty. Similarly, the DF range in the negative region would be restricted if the load ratio were much greater than 2:1. Also, it is felt that the ratio should be no greater than necessary, because of the stability problem.

It might seem that the chart could have been split between two curves, one positive and one negative, each optimum for its own region. But this does not work out well. For one thing, it is evident that two curves will be necessary anyway in the negative region in order to depict it properly, and, it should be noted, Scale A works out to be just about right for the middle- and low-negative region scale. This way, with one curve running through both regions, the chart gives to the user a much more meaningful picture of the positive and negative damping factor relationships, and further permits the measurement of most practically encountered damping factors with only one set of terminating resistors.

Practical Accuracy

At first glance, the setting up of two loading conditions wherein one is only 10 per cent higher than the other, as is the case with Scale B, might seem to be an invitation to serious error, but it is not.

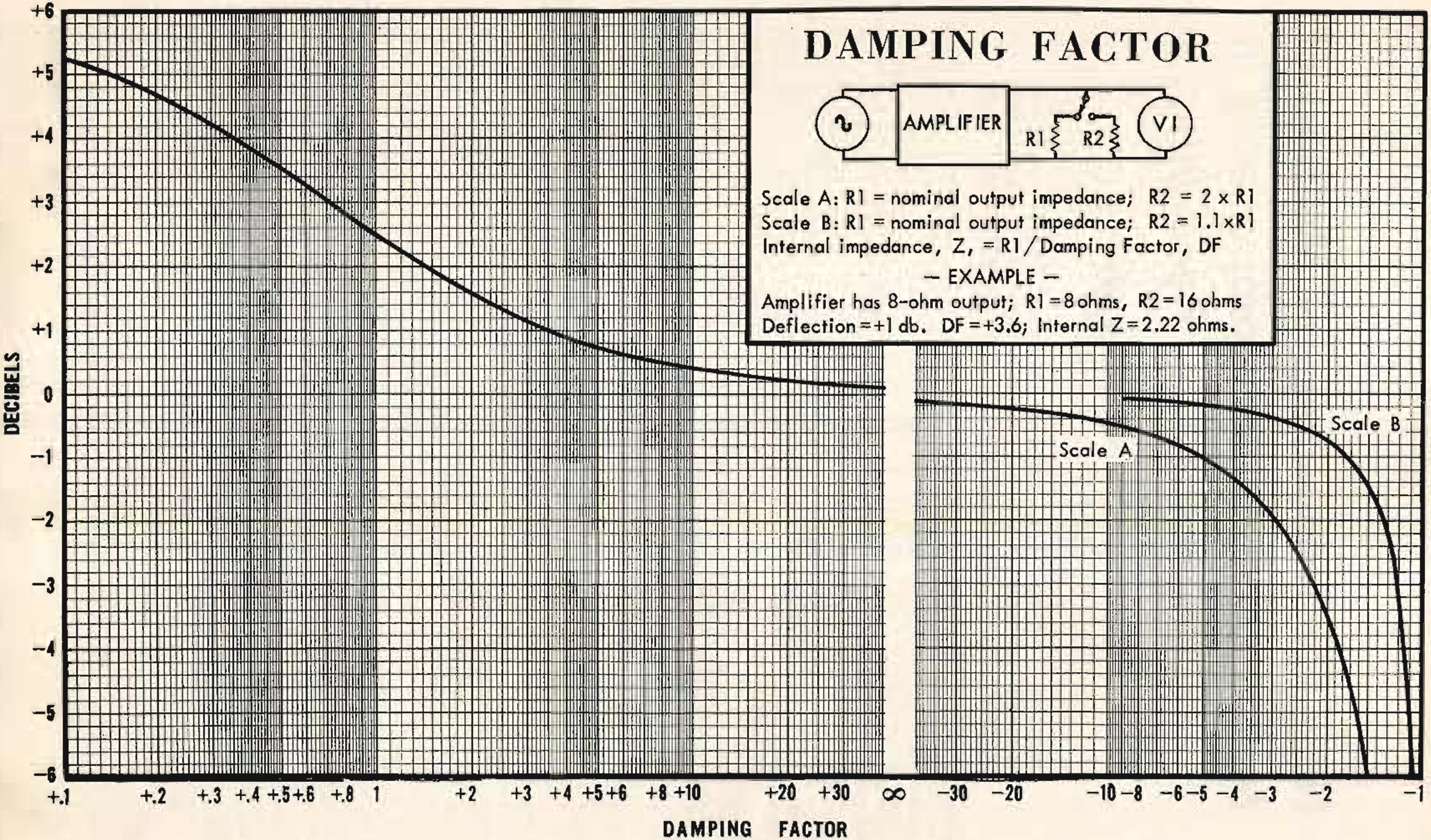
Assume the worst that can happen:

(*Continued on page 78*)

* *228 S. Summit, Iowa City, Iowa.*

[1] F. Langford-Smith, "Letters," AUDIO, July, 1955.

[2] Joke.


DAMPING FACTOR
AMPLIFIER
R1
R2
VI
Scale A: R1 = nominal output impedance; R2 = 2 x R1
Scale B: R1 = nominal output impedance; R2 = 1.1xR1
Internal impedance, Z, = R1/Damping Factor, DF
— EXAMPLE —
Amplifier has 8-ohm output; R1 = 8 ohms, R2 = 16 ohms
Deflection = +1 db. DF = +3.6; Internal Z = 2.22 ohms.
Scale A
Scale B
DECIBELS
+6
+5
+4
+3
+2
+1
0
−1
−2
−3
−4
−5
−6
+.1
+.2
+.3
+.4
+.5
+.6
+.8
1
+2
+3
+4
+5
+6
+8
+10
+20
+30
∞
−30
−20
−10
−8
−6
−5
−4
−3
−2
−1
DAMPING FACTOR

New Electromechanical Method of Matrixing the Two Components in Stereophonic Disc Recording

HORST REDLICH,* HANS-JOACHIM KLEMP,** and STEPHEN F. TEMMER***

While the 45/45 stereo cutter can be fed directly from the right and left channels without matrixing, it is equally possible to employ a 0/90 cutter—when designed especially for the operation—to obtain exceptionally fine results.

For the longest time efforts have been directed towards the ability to store stereophonic sound. In magnetic systems, such as tape or film, a satisfactory solution has been found by dividing the available magnetic track width. The modern phonograph record, however, is one recording medium which already has the highest known ratio of playing time to available surface area. Approximately 30 seconds of sound can be stored on one square inch of its surface, while tape at 3¾ ips *twin-track*, requires at least *fifteen* times the surface area to accommodate the same amount of sound. This is one of the reasons why it is nigh onto impossible to apply principles analogous to magnetic recording to disc. As a result, early attempts at recording two channels on disc approached the problem not by dividing the track width, but rather by cutting the available surface area in half.

The Double-Groove Principle

There have been, for example, disc recordings (Cook Laboratories) which are so recorded as to separate the two recording channels into two separate groups of normal microgrooves situated next to each other on the same side of the record. This type of disc must be played back using a double tone arm with two separate pickup cartridges.

This method has two distinct disadvantages. The first is the demand for double the surface area needed for monophonic recordings, and the resulting short playing time available per record. The second, and even more serious, is the extreme difficulty of maintaining the necessary phase synchronism between channels. The degree of accuracy of the phase relationship between the two channels, is dependent on the alignment of the recording and reproducing styli. Aside from this, the phase alignment of the two channels must remain fixed from the outermost to the innermost groove of the record, requiring the same tangential or arc-type motion for both the recording and reproducing mechanism.

All of these problems would be avoided if it were possible to use a single groove, as in monophonic recording, which is recorded with a single stylus and is played back with a single stylus. There are two possible solutions to the problem of how two separate signals can be accommodated in such a single groove.

The Carrier-Frequency Method

This is essentially a method similar to that employed in carrier telephony. Here is one type of carrier system: The first channel is fed directly to the recording cutter through a high-pass filter network of, for instance, 12 kc. A carrier frequency lying directly above the first channel is then used to modulate the contents of the second channel. The upper side-band ($f_c + f_{ch2}$) is filtered out and is recorded together with channel 1 and a synchronizing frequency. The recorded band width, therefore, is somewhat more than twice the width of each channel alone. With a frequency range of 12 kc in each channel, a recorded bandwidth of about 25 kc is required.

The reproduction requires a playback system capable of an upper frequency limit of 25 kc. After preamplification, the first channel is filtered out by means of a low-pass filter. A high-pass filter then isolates channel two, which is demodulated in a ring modulator by addition of the carrier frequency. This frequency is generated by a multivibrator which is triggered by the synchronizing pulse originally recorded on the disk. This synchronizing pulse is necessary in order to achieve flawless demodulation even when the turntable speeds of the recording lathe and reproducing turntable do not agree.

The above described system was developed over a long period of time in the Laboratories of English Decca. Despite the difficult demands on the bandwidth of the recorded disc, the Decca engineers had achieved a very excellent solution to the problem of two-channel recording in a single groove. They achieved a reproducing quality in every way equal to today's highest accomplishments in the electroacoustical field.

The major disadvantage in this system is the relatively high equipment contribution which the consumer must make to achieve demodulation. It is for this reason that another solution to the problem of two-channel, mono-groove recording was sought.

The Two-Component or Vector System

While the foregoing method achieves single-groove recording and channel separation by electronic means, a mechanical alternative to this is possible. The recorded groove contains the two information channels as two excursions lying at right angles to each other (orthogonal system). The actual orientation of this set of crossed motions with respect to the surface of the disc is immaterial, however practical considerations make two particular orientations preferential: one vertical-lateral; the other in which each channel is oriented at 45 deg. to the surface of the disc. The resulting groove may be compared to a street which runs over hill and dale while at the same time winding and curving. The motion of a car on such a road may be analyzed by separation into two components lying at right angles to each other, which may then be in turn assigned to channel one or two.

The recording of such signals requires a transducer capable of translating the signals of channels one or two into excursions of the stylus in the proper

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*** *President, Gotham Audio Development Corp., 2 W. 46th St., New York, N. Y.*

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direction. The reproducing transducer does the reverse. The complex mechanical motion of the reproducing stylus is resolved into excursions lying at right angles to each other which are then converted into electrical impulses. Of utmost importance is the accuracy of the angle which the "motional cross" has to the surface of the disc in both recording and reproduction, and the true perpendicular relationship which the components have to each other. Only in this manner can maximum separation between channels be achieved.

The first experiments with this type of stereophonic recording were made by A. D. Blumlein, engineer with E.M.I. Ltd., England. Since these experiments and his patent go back to the early thirties, it must be ascertained how this type of recording is adaptable to the present-day microgroove recording techniques. Experiments of TELDEC (Telefunken-Decca) go back to the years 1954 and the first cutterhead utilizing feedback in both channels was built in the summer of 1955.

Quality Considerations in the Vectoral System of Recording

Provided we assume the necessity of preserving the full stereophonic effect in two channels, each of which are of equal quality, then each of them should have a degree of quality, with respect to frequency response and distortion equaling the highest attained level of today's electroacoustical knowledge. Furthermore, we must aim at the highest degree of interchannel separation. The degree of interchannel separation necessary to preserve the full sound-width of the original is subject to much theorizing and discussion. We have conducted numerous listening tests in an effort to arrive at some reasonable figure and on the basis of these tests we feel that the high quality of a stereophonic recording can only be maintained if the separation between channels stays above 20 db. Both the original recording on tape as well as the disc mastering and reproduction separations are included in this figure. The recording alone should be considerably better than the 20 db quoted, in order to allow for a greater tolerance spread in mass-produced reproducing equipment. A separation figure of more than 30 db should be the aim of the disc recording industry. This degree of separation requires of the cutting head a very high degree of motional accuracy. To translate this into concrete figures: the deviation from the prescribed two paths of the stylus by all influences may not exceed 1.5 deg. At higher frequencies, as for instance 10 kc, this is an actual physical displacement of only 0.001 mil. Such magnitudes can no longer be measured by mechanical means. In other words, we demand of a mechanical system degrees of accuracy which cannot be controlled by mechanical or optical means. The only possible approach in our development of such a cutter was to utilize a system of electromotional feedback for the stabilization of stylus motion. This type of feedback control also made possible the other required quality demands for wide frequency response and low distortion, which are favorably influenced by motional feedback. There are, however, further problems of non-linear distortion in such a two-component system, which will be covered in a later paragraph.

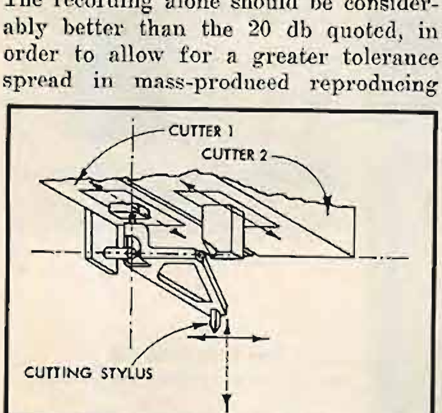

Fig. 1. Mechanically coupled two-component cutterhead.

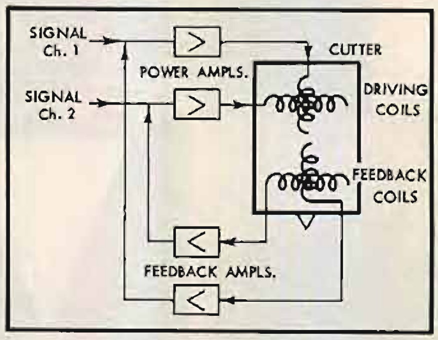

Fig. 2. Diagram of feedback and driving coils in the TELDEC cutterhead.

A Double-Motion Feedback Cutting Head

The integration of the individual components into a resulting double-motion of the recording stylus can be accomplished in two basic ways.

The first way is to use two normal single-motion transducers connected through a lever system as shown in *Fig.* 1. The motion of system 1 leads to motion of the stylus in a vertical direction, while motion of system 2 produces a lateral excursion. Motion of both systems simultaneously produces a vectoral resultant of the two motions, whose direction and amplitude is the resultant of the applicable amplitudes of the two directions as well as their phase relationship.

Experiments carried on with such a system demonstrate its feasibility, as shown by a similar type of drive found in the Westrex Type 3B cutterhead. The required degree of accuracy of motion is, however, extremely difficult for such a system to attain, *even* if the quality and stability of the individual motional components are of the *highest accuracy.* The motional feedback applied to each of such systems cannot include the stylus suspension system itself due to the lever-type connection between the systems and the stylus, whose self-generated interference cannot be compensated. It seemed therefore desirable to devise a cutter in which the armature is of extreme stiffness in itself, and which is so suspended as to permit free motion in only one plane. Only in this way is it possible to use two feedback systems which interact to produce stabilization of motion between the two channels.

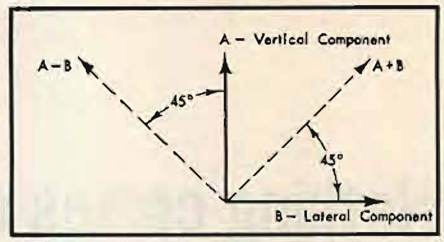

Fig. 3. Transformation of a vertical-lateral recording into a 45/45 recording by sum and difference formation.

The feedback loops are so arranged as to produce at the same time a complete feedback system within each motion (*Fig.* 2). The feedback voltages are produced in two separate feedback coils situated near the stylus itself, and are fed back to the appropriate cutting amplifier. This kind of arrangement produces a linearity compensation within the two loops, and furthermore serves to negate any mechanical outside influence which would tend to displace the stylus motion from its assigned path. Let us take an example where a mechanical resistance diverts the stylus from its intended motional direction. This produces a voltage in the *opposite* feedback loop which is proportional to the diversion. Since this feedback voltage is applied with reversed polarity through the driving amplifier to the drive coil placed at right angles to it, a motion equal and opposite to the mechanical diversion is created which offsets it and maintains linearity.

The result of such a feedback system is the assurance that the two motions of the stylus are maintained accurately perpendicular to each other. Such perpendicularity is absolutely essential, for only in this way can the minimum degree of interaction between channels be assured. The relative position of the "motional cross" should be either vertical-lateral, or at 45 deg. to the disc surface. The selection of one or the other system simply requires a different placement of the driving and feedback coils on the armature of the cutter. It is also possible simply to "turn" the "motional cross"

(*Continued on page 83*)

Ralph Bellamy, starring in "Sunrise At Campobello", listens to stereo on his Collaro changer and Goodmans Triaxonal Speaker System.

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Understanding db, dbm, and VU

JULIAN L. BERNSTEIN*

An interesting and informative clarification of the differences between the terms used in describing power and voltage ratios, together with the required formulas and instructions for their use.

Quite early in the history of the telephone, engineers recognized a need for a special unit with which to measure and compare the losses of different telephone lines. For various reasons, they selected the loss of a mile of "standard cable" at 1000 cps as their basic unit. This became known as the "transmission unit," and later on, as the decibel. Today, the decibel has taken on added significance, as a glance at any technical advertisement will show, for it is common practice to specify equipment performance in terms of decibels. A good working knowledge of the decibel, and its first cousins, the dbm and the VU, is essential in order to understand todays technical literature thoroughly. It is the purpose of this article to aid in supplying such understanding, and clear up certain misconceptions regarding these units.

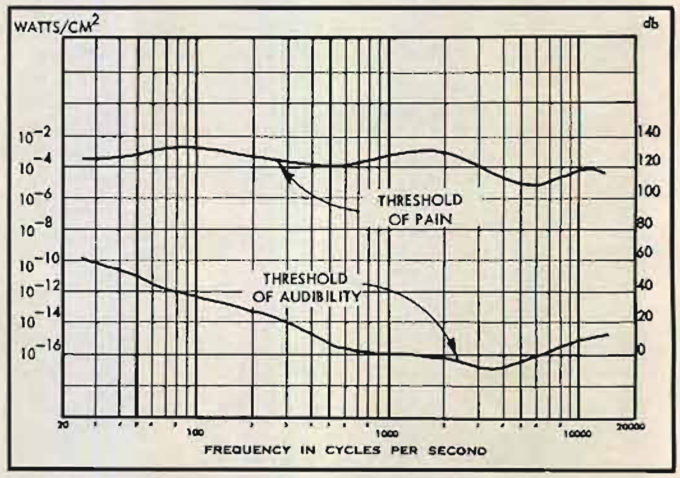


Fig. 2. Thresholds of audibility and of pain. Note that at 1000 cps the pressure difference is 10^{12} between the thresholds, but the db difference is only 120. (Courtesy Bell Telephone Laboratories.)

The Decibel

In electrical circuits, emf, current, and power are customarily measured in units named volts, amperes, and watts respectively. Although these units may be used for measuring electrical audio signals, it is advantageous to specify a different unit for such signals, one which is based on human hearing. The unit should also be independent of the system in which the measurement is to be made, that is, it should function equally well for an acoustic pressure wave as for its electrical equivalent. The *bel* (named after Alexander Graham Bell, inventor of the telephone) is such a unit, and is defined as the logarithm (to base 10) of the ratio of the power content of the wave being measured to that of another wave, known as the reference. Algebraically, the bel is found from the relationship

$$N_b = \log \frac{P_1}{P_2}$$

in which N_b represents the number of bels being produced by the wave being measured, P_1 is the power content of that wave, and P_2 is the power content of the reference wave.

In practice, the bel is much too large a unit, so it is divided into tenths, each called a *decibel*, abbreviated *db*. Dividing by 10 means that there will be ten times as many smaller units in each bel, causing the equation above to change into

$$N_{db} = 10 \log \frac{P_1}{P_2} \qquad (1)$$

Observe that in Eq. (*1*), P_1 and P_2 both represent power. Thus the units cancel, leaving the ratio a dimensionless quantity, thereby permitting the decibel to be easily applied to any wave, acoustic or electric.

In Eq. (*1*), the ratio of P_1 to P_2 may be any value. In the event P_1 is the larger of the two, the ratio is greater than unity and the resulting value of N_{db} will be positive. On the other hand, should P_2 be the larger, N_{db} acquires a negative sign, since the logarithm of a number less than 1 has a negative characteristic. When working with ratios less than unity, two important considerations must be taken into account. The first of these is that a negative value of N_{db} *does not* signify a reversal of direction, as does a negative voltage or current in an electrical circuit. The only significance of negative db values is that they indicate the measured wave is smaller in power content than the wave to which it is being compared. The second consideration is purely mathematical and involves the correct notation for a negative logarithm characteristic. For instance, it is customary in algebra to express the log of a small number, say .002, as 7.301 − 10. For decibel notation, this approach is exceedingly clumsy and should be avoided. Instead, the indicated subtraction should be made immediately. Thus, for the number above, the positive portion of the logarithm (7.301) should be added to the negative portion (−10), resulting in −2.699 as the answer. This latter value has the virtue of being much easier to work with, and is just as valid as the original logarithm written in the usual manner.

Decibel notation, as the equation indicates, is *always* a comparison between two values. There are therefore many instances in which N_{db} will appear to be the same numerically although the individual power values are not necessarily identical. To illustrate, suppose that the power dissipated in a certain resistor is 1 watt. Now, suppose that the power is increased, say to 2 watts. The increase

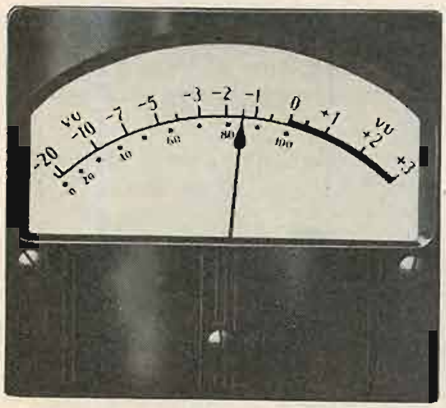
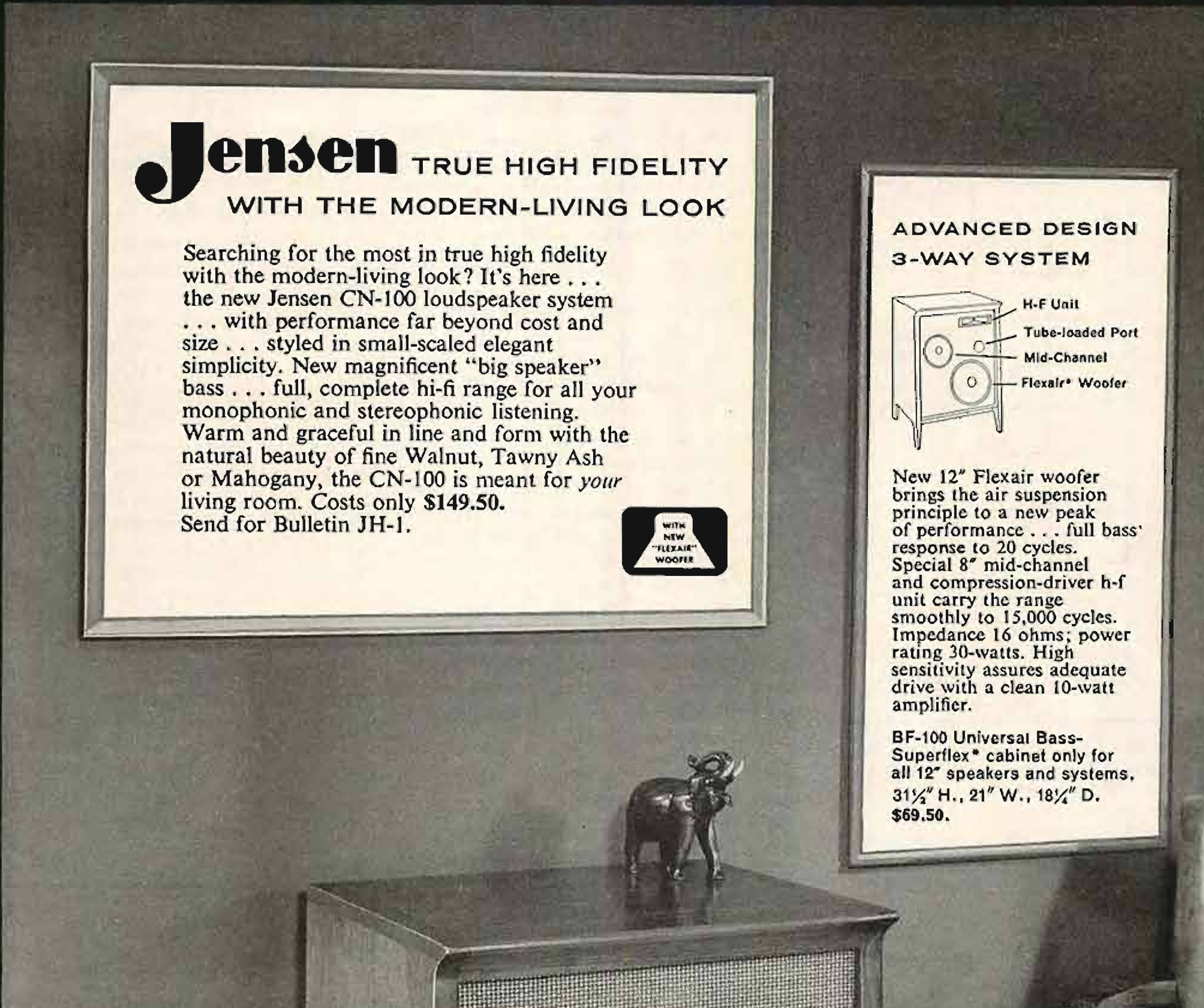


Fig. 1. A typical VU meter scale, used to compare amplitudes of audio signals. Note that 0 appears approximately 2/3 of the way up the scale. This corresponds to 0.775 volts applied across the meter and its internal rectifier. (Courtesy Weston Instruments Division of Daystrom, Inc.)

* *RCA Institutes, Inc., 350 W. 4th St., New York 14, N. Y.*



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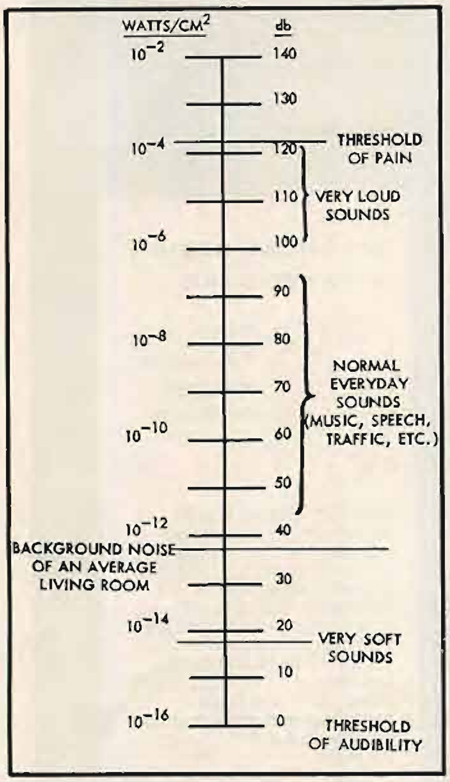

Fig. 3. Decibels vs. acoustic power using the threshold of audibility at 1000 cps as 0 db.

in power, in terms of db, may be found by merely substituting the values in Eq. (*1*), thus:

$$N_{db} = 10 \log (2/1) = 3.01 \text{ db}$$

But consider what would happen if the original power dissipation had been 5 watts. Doubling the power would cause 10 watts of dissipation to appear in the resistor. Again substituting values into the equation,

$$N_{db} = 10 \log (10/5) = 10 \log 2 = 3.01 \text{ db}$$

From this it is clear that whenever the power in a circuit is doubled, it is the equivalent of an increase of 3.01 db above the original value. Oddly enough, this characteristic of the decibel closely parallels the psychology of human hearing, for the ear judges sounds by comparisons, comparing what it hears now to what it heard a moment ago. Thus, if a sound is made twice as loud as previously, regardless of its original value, the ear interprets this as a 3.01-db increase. Conversely, cutting the power in half, which is the equivalent of inverting the ratio in equation 1, causes N_{db} to become −3.01. This value (3.01 db), which is simply referred to as 3 db for convenience, is very important in audio circuits, for the average human ear recognizes 3-db changes, either increases or decreases, as a definite change in the loudness of a given sound, while it is almost insensitive to 1-db changes, and unless trained by practice, cannot recognize 2-db changes. It is for this reason that bandwidth is usually measured between the two frequencies (called the half-power frequencies) at which the output power is 3 db less than at the center of the band.

Extending the range of ratios shows that if the power ratio is quadrupled, $N_{db} = 6$. For a power ratio of 10 to 1, $N_{db} = 10$, while a ratio of 100 to 1 makes N_{db} equal to 20. Continuing on up the ratio ladder, it is easily seen that *multiplying* the ratio of P_1 to P_2 by 10 is the same as *adding* 10 units to N_{db}. Thus, for a 1000 to 1 ratio, N_{db} becomes 30, and for a million to one ratio, $N_{db} = 60$. Again, this coincides with human hearing, for the ear drum moves only 120 times as far for the loudest sounds (threshold of pain) as it does for the softest sounds (threshold of audibility) in the vicinity of 1000 cps. This ear motion, however, corresponds to an acoustic power ratio of somewhat more than 1,000,000,000,000 to 1, as shown in *Fig.* 2. For such a ratio, N_{db} is slightly greater than 120. Clearly, an additional advantage appears in using the db, for it permits small numbers to express large values. Because of this reduction in numerical size, the db is often referred to as a *compressed* unit.

Power Reference Levels

Regardless of the value of P_1 in Eq. (*1*), if it is equal to P_2, N_{db} must equal zero. Electrically, zero implies the absence of the quantity being measured, and ideally, 0 db should have this meaning. This is not possible, however, for the db is based on a logarithmic quantity, which can only equal 0 when the ratio equals 1. Because of this, some arbitrary value of power must be selected to represent 0 db, with this chosen value being known as the *reference level*. Any other value of power appearing in a circuit may then be expressed in db *with respect to the reference*. For measurement purposes, P_2 is always considered to be the reference, while the power content of the wave or signal being measured is represented by P_1. The resulting value of N_{db} is often known as the *power level* of the wave under measurement.

The reference level, as stated above, is arbitrarily selected. Logically, however, it should correspond to the threshold of audibility, and acoustically, this is the value so used. *Figure* 3 illustrates the resulting power levels for various values of acoustic power based on this reference. For radio, TV, and recording circuits this reference turns out to be too small, for most circuits are designed to operate with loudspeakers that must provide sounds substantially higher than the threshold of audibility in order to overcome background noises, which average around 35–40 db, as the figure indicates. Therefore, for electrical circuits, P_2 has been standardized at 6 milliwatts of power dissipated across 500 ohms resistance, which is usually written as 6 mw/500 ω. *Figure* 4 shows the resulting power levels for electrical circuits using a 6 mw reference. It is interesting to note that this ladder is symmetrical around 0, for a given ratio, if inverted, provides the same change in N_{db}.

The VU and the dbm

If all audio signals were pure sine waves, the db would be the only unit required for measurement purposes. Unfortunately, this is not the case, and most audio signals are quite complex. This in turn causes difficulty in measuring their relative strength in db, simply because the rms amplitude of a complex wave is not 0.707 times the peak value. In fact, for some signals, the rms value may be as small as .05 times the peak. This being so, broadcast engineers set out to design a meter that would accurately compare the relative amplitudes of various audio signals. It was found that it would be more convenient to define a new unit for this meter to measure, rather than to rely on the overworked decibel. Since the meter was to measure what might loosely be termed volume, what could be more natural than to call the new unit a *volume unit*, abbreviated *VU*. The VU is similar to the decibel since it uses a logarithmic ratio, but it

(*Continued on page 89*)

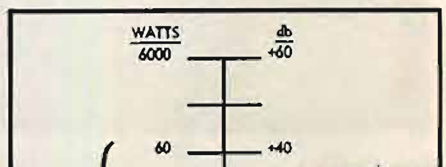

Fig. 4. Decibels vs. electric power using 6 mw as 0 db. Based on this reference, most commercial amplifiers are capable of producing an output of +30 to +40 db.

RC Filter Design for High-Impedance Crossover Networks

CHARLES W. HARRISON, Jr.,
Cdr., USN

The techniques and procedures of high-impedance RC crossover network design are simple once they are understood. The author gives design criteria for two- and three-section filters in the form of working formulas.

Two or three section RC dividing networks are frequently employed in dual channel amplifier systems.[1,2] These networks should permit uniform transition of the delivery of power from the bass speaker to the treble speaker and vice-versa in the region of crossover. To achieve this the sum of the powers in the two channels must be constant with respect to frequency for a signal voltage of constant amplitude applied to the input terminals of the filters. The filters discussed in this paper permit this to be accomplished to a high degree of approximation.

The origin of design formulas for high-impedance RC crossover networks is a mystery to the average audio hobbyist. For this reason he is reluctant to change circuit parameters in a standard filter design for fear of obtaining a degradation in over-all performance. But once the basic principles of design are understood the experimenter will realize that an infinite number of RC filters for crossover application will meet his requirements. He will be able to determine circuit parameters, with facility, for two-, three-, or more, section filters. The purpose of this article is to permit the hobbyist to "see through a glass" more clearly with respect to RC filter design.

Part 1. DERIVATION OF APPROXIMATE FORMULAS

1. Three-Section Low-Pass Filter

A low-pass three-section RC filter is shown in *Fig.* 1. Each section, consisting of a resistor R_n and capacitor C_n connected in the circuit as illustrated, is consecutively numbered from left to right. It is assumed that section 1 is driven by a cathode follower of low internal impedance, and that section 3 works directly into the grid of the inupt tube of the bass amplifier.

The design is based on the following criteria:

(a) Each filter section is to provide 1 db of voltage attenuation at the crossover frequency. This requires the time constants of each filter section to be equal.

(b) No filter section is to load the driving filter section appreciably; i.e., section 2 is not to load section 1 significantly, etc., at any frequency. (It is tacitly assumed in these calculations that each section works into an open circuit.)

(c) The entire filter and the succeeding vacuum-tube circuit is not to load the cathode follower unduly at any frequency.

If E_{in} and $E_{1_{out}}$ are the input and output voltages of section 1,

then

$$db = 20 \log_{10} \left| \frac{E_{in}}{E_{1_{out}}} \right|_{f = f_c} = 1 \quad (1)$$

or

$$\left| \frac{E_{in}}{E_{1_{out}}} \right|_{f = f_c} = 10^{0.05} = 1.12 \quad (2)$$

Here f_c is the crossover frequency. An inspection of the circuit shows that at any arbitrary frequency f,

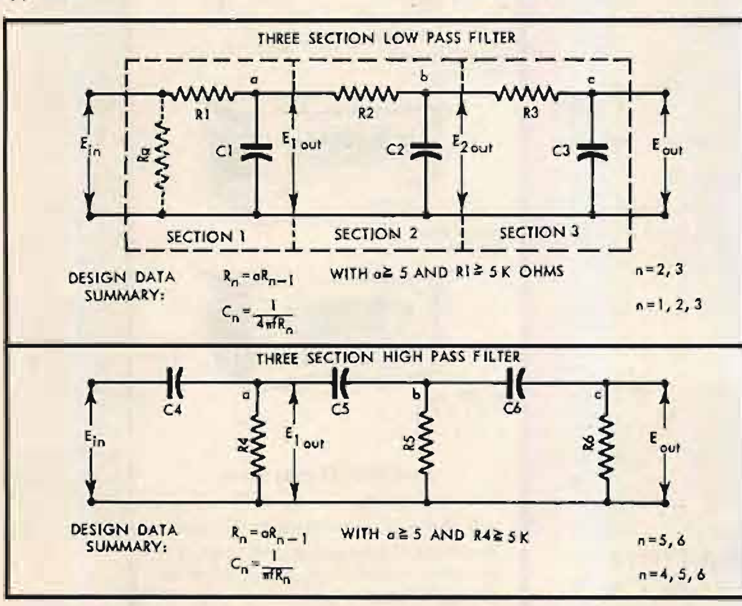

Fig. 1 (left). Three-section low- and high-pass filters.

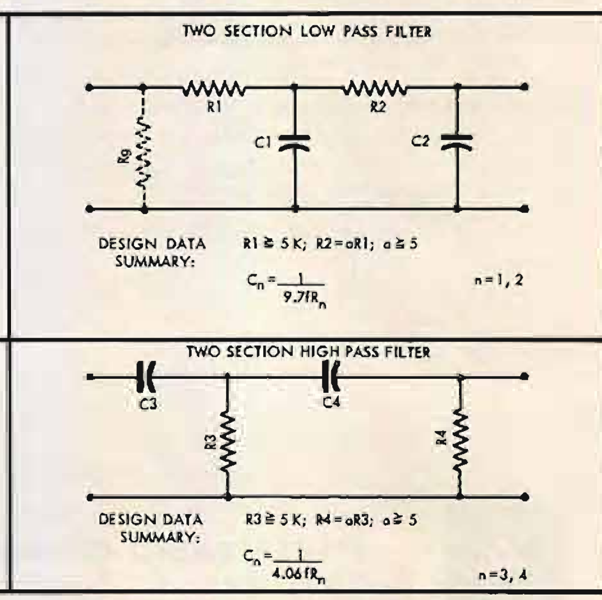

Fig. 2 (right). Two-section low- and high-pass filters.

* *9700 Snow Heights Blvd., N.E., Albuquerque, N. M.*

[1] Benjamin B. Drisko and R. D. Darrell, "40-db Feedback audio amplifier." *Electronics*, March 1952.

[2] Charles W. Harrison, Jr., "High-quality dual-channel amplifier." Audio, January, 1956.

$$\left|\frac{E_{in}}{E'_{out}}\right| = \left|\frac{R_1 - j/\omega C_1}{j/\omega C_1}\right| \qquad (3)$$

In Eq. (*3*), R_1 and C_1 are the resistive and capacitive elements of section 1 and $\omega = 2\pi f$. When $\omega = \omega_c$ one may equate the magnitude of Eq. (*3*) to (Eq. (*2*). Thus

$$\sqrt{\omega_c^2 R_1^2 C_1^2 + 1} = 1.12 \qquad (4)$$

or

$$\omega_c R_1 C_1 = \frac{1}{2} \qquad (5)$$

Hence,

$$C_1 = \frac{1}{4\pi f_c R_1} \qquad (6)$$

At high frequencies the reactance of C_1 will be small (point *a* will be at ground potential) so that R_1 is the cathode-follower load. To satisfy design criterion (c), R_1 should be at least 5000 to 10,000 ohms. Furthermore, design condition (b) requires

$$R_3 = 5R_2 = 25R_1 \qquad (7)$$

or better yet,

$$R_3 = 10R_2 = 100R_1 \qquad (8)$$

In either case it is worth noting that R_2 is the geometric mean between R_1 and R_3. Suppose one selects $R_1 = 10{,}000$ ohms. Then from Eq. (*8*), $R_2 = 100{,}000$ ohms and $R_3 =$ 1 megohm. Since R_1 is specified, C_1 is determinable from Eq. (*6*). Design condition (a) requires

$$R_1C_1 = R_2C_2 = R_3C_3 \qquad (9)$$

so that

$$C_2 = \frac{R_1C_1}{R_2} \qquad (10)$$

and

$$C_3 = \frac{R_2C_2}{R_3} \qquad (11)$$

If the input capacity of the bass amplifier is significant, C_3 as computed from Eq. (*11*), should be reduced appropriately.

C_2 is the geometric mean between C_1 and C_3, if R_n is chosen as indicated in Eq. (*7*) or Eq. (*8*).

An inspection of the circuit reveals that the low-pass filter does not provide a grid return. The grid leak should not be connected across the output terminals of section 3 for reasons of loading. Instead, a resistor of appropriate value R_g is connected across the input terminals of section 1. Then $R_1 + R_2 + R_3 + R_g$ forms the grid return of the input tube in the bass amplifier.

This completes the design of the low-pass three-section filter. A summary of design data is given in *Fig.* 1. It is worth mentioning that the larger the value of the multiplier *a* in the formula $R_n = a\,R_{n-1}$ the more accurate the expression for C_n becomes. The judicious choice of resistor values may permit use of capacitors of standard value.

2. Three-Section High-Pass Filter

A three-section high-pass filter, composed of resistors R_n and capacitors C_n, is also pictured in *Fig.* 1. It is assumed that the voltage E_{in} is delivered by the cathode follower and that E_{out} is the voltage applied to the input tube of the treble amplifier. R_6 is the grid resistor. The design criteria are the same as stated for the low-pass filter.

$$\left|\frac{E_{in}}{E_{out}}\right|_{f=f_c} = 1.12 = \left|\frac{R_4 - j/\omega_c C_4}{R_4}\right| \qquad (12)$$

Then

$$\frac{\sqrt{\omega_c^2 R_4^2 C_4^2 + 1}}{\omega_c C_4 R_4} = 1.12 \qquad (13)$$

Solving (*13*) for C_4, one obtains

$$C_4 = \frac{1}{\pi f_c R_4} \qquad (14)$$

At some high frequency the reactance of C_4 becomes small, so that the cathode-follower load approximates R_4 (it is assumed that $C_4 \geq 5C_5$). Accordingly, R_4 should be 5000 to 10,000 ohms. If $R_4 = 10{,}000$ ohms, C_4 is calculable from Eq. (*14*). From design criterion (a) one knows that the time constants of all filter sections must be equal if each section is to provide 1 db of attenuation. Therefore,

$$R_4C_4 = R_5C_5 = R_6C_6 \qquad (15)$$

Also, to satisfy design criterion (b),

$$R_6 = 5R_5 = 25R_4 \qquad (16)$$

as minimum. Preferably,

$$R_6 = 10R_5 = 100R_4 \qquad (17)$$

The design of the three-section high-pass filter is now complete. A summary of design information is provided in *Fig.* 1. Again, the larger the value of *a* in the relation $R_n = a\,R_{n-1}$, the more accurate the formula for C_n becomes.

3. Two-Section Low- and High-Pass RC Filters

The same techniques and procedures employed in the design of three-section filters are applicable in solving for circuit values for use in two-section crossover networks. *Figure* 2 presents necessary design information for the low- and high-pass filters. It is assumed that each filter section must provide 1.5 db signal attenuation at the cross-over frequency.

Illustrative example:

Design a two-section 800-cps high-impedance crossover network to be driven by a low-impedance cathode follower. (When determining filter circuit parameters it is convenient to have at hand a table of resistor and capacitor values manufactured as standard items.)

Refer to *Fig.* 2. Select $R_1 = 13{,}000$ ohms and $R_2 = 130{,}000$ ohms. Then

$$C_1 = \frac{1}{9.7 \times 800 \times 13{,}000} \approx 0.01 \times 10^{-6} \text{ farads} = 0.01 \text{ mfd.}$$

$$C_2 = \frac{C_1R_1}{R_2} = 0.001 \text{ mfd.}$$

Let $R_3 = 15{,}000$ ohms and $R_4 = 150{,}000$ ohms. Then

$$C_3 = \frac{1}{4.06 \times 800 \times 15{,}000} \approx 0.02 \times 10^{-6} \text{ farads} = 0.02 \text{ mfd.}$$

$$C_4 = \frac{R_3C_3}{R_4} = 0.002 \text{ mfd.}$$

Since the filters are driven from the same source, the maximum possible load on the cathode follower (assuming that each filter works into an open circuit) is the parallel combination of R_g, R_1, R_3 and R_4. If the minimum load is to be 5000 ohms, R_g, the grid return for the input tube of the bass amplifier may have the value $R_g = 20{,}000$ ohms. The total grid resistance is then 163,000 ohms.

A two-section crossover network has been built according to the above specifications; however, resistors and capacitors of ±10 per cent tolerance were employed in the construction. The actual component values were not measured nor was any "padding" of the completed network attempted. The results of measurements on this network are shown in Fig. 3.

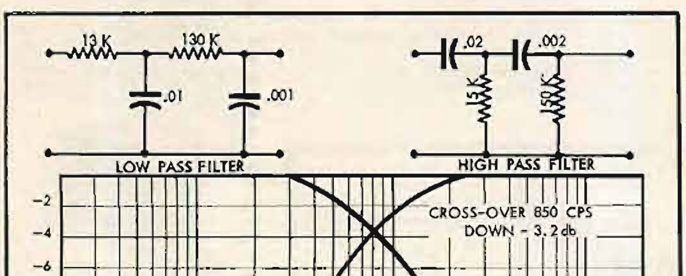

Fig. 3. Performance data of two-section crossover network.

(*Continued on page 66*)

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Design of a High Quality Stereo Console

R. A. JOSS*

"Handsome is as handsome does" it is said; but even if this installation were not handsome in performance, which it should be considering the equipment employed, it is certainly attractive enough for any home.

With the current hi-fi "boom," a number of very expensive package consoles are coming on the market, all highly touted as the last word in sound reproduction for the home. To the initiate, many of these units lack the wide range necessary for quality reproduction, but they *do* offer the "advantage" of housing everything in one handsome package, and, let's face it, many many home music systems are sold on the basis of appearance rather than performance.

As hi-fi specialists, and with the above in mind, we decided to see if it were possible to design a unit using standard high-fidelity components which would offer the convenience of a package unit with the high-quality sound of true wide-range reproduction. While we were at it, we decided that the "monster" would incorporate "all modern conveniences," with built-in tape recorder and stereophonic playback. It was also decided to include both record changer and a transcription turntable, the latter designed only for single-speed operation. The changer provides non-critical background music listening, and the additional $16\frac{2}{3}$, 45 and 78 rpm speeds, while the simplification of the turntable permits the choice of a high-quality but low-cost unit.

* *Sound Scription Service Inc., 5239 Park Ave., Montreal, P.Q., Canada.*

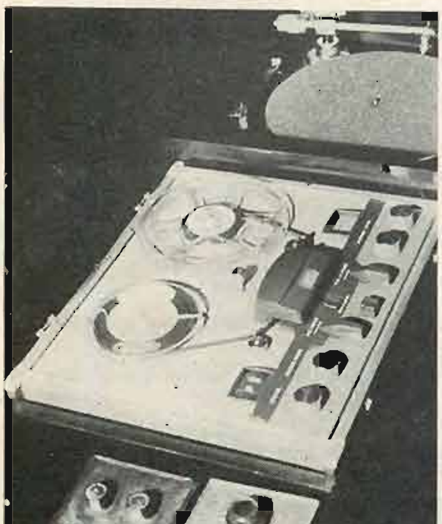

Fig. 1. View of the "well" containing portable Ampex A-122 tape recorder.

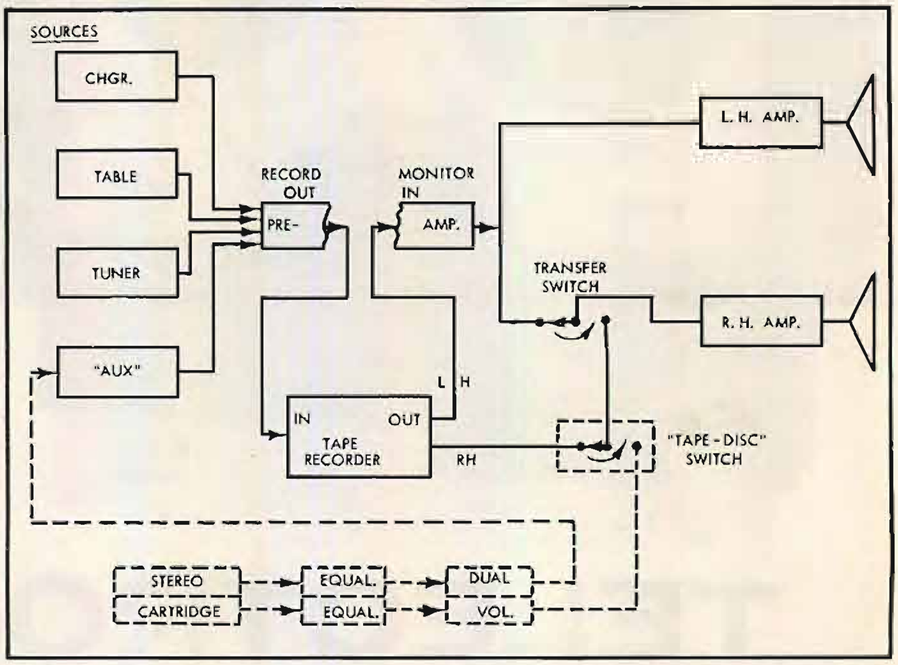

Fig. 2. Block diagram showing interconnection of components. Dotted lines indicate proposed future additions.

Acoustic Feedback

It was realized that the major problem to overcome would be that of acoustic feedback between the loudspeakers and the record players. Most package units deliberately restrict the bass range to avoid this problem, but this was to be a no-compromise system, with extended bass response. Various ideas were tried and discarded, but the solution turned out to be quite simple. The basic solution to the problem of acoustic feedback is to isolate the speakers from the record-player. Usually distance is the isolating medium, but with the advent of the acoustic suspension technique, truly wide-range reproduction became available from relatively small enclosures, which could be acoustically isolated from the cabinet containing the turntables by "floating" the enclosures on a foam-rubber pad inside the equipment cabinet. This relatively simple technique proved entirely satisfactory—the turntable and arm is directly over one of the AR-1's, and no trouble whatever has been encountered with acoustic feedback.

Choice of Electronic Components

Driving the speaker systems, we have two 30-watt amplifiers, but the preamp was something of a problem. No stereo preamps with all the required features seem to be available as yet on the market. After a study of specifications, a Bogen PR-100A was chosen because of its great flexibility, although other quality preamps could be adapted for the job. Since operating simplicity was considered of some importance, it was decided to use only one preamp, and sacrifice tone controls on the right channel when playing from stereo sources. The tape recorder selected incorporates dual preamps with

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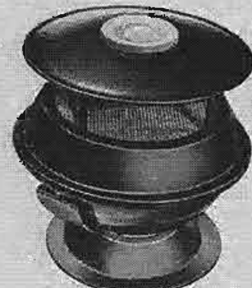

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WHY THE STEREOSFERE FOR STEREO!

It is well known that the ear cannot detect the accurate position of the source of sound below approximately 300 cycles. Therefore, frequencies below 300 cycles from both amplifiers in a stereo system, can be combined and fed to a single loudspeaker without any loss of the stereo effect. Because of this an existing full range speaker unit, properly baffled, needs only one STEREOSFERE to complete a fully balanced stereo speaker system for an average-sized living room or monitoring studio. Result is substantial savings for you, as well as a much more modest-sized second speaker system.

The STEREOSFERE is designed to reproduce all audio frequencies from 300 to 20,000 cycles. One STEREOSFERE combined with an existing full range system makes an excellently balanced stereo speaker system. The model S10-30 STEREOSFERE, speaker will handle 30 watts of continuous program material. The IDM-30 frequency-dividing, isolation and mixer network, is specifically designed to handle a maximum of two S10-30 STEREOSFERE speakers. When so used the existing full range speaker system handles only the combined bass signals. Above 300 c/s the channels are fed one to each S10-30 STEREOSFERE. The *stereo balance* is then excellent, because the high frequency units are identical.

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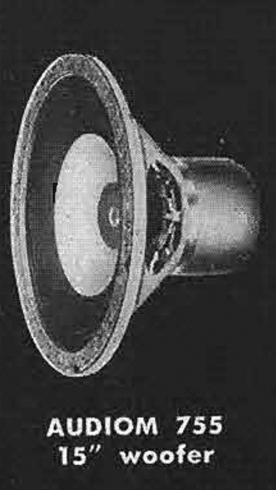
*The Goodmans *Rigidflex* cone has a completely flexible free floating cone rim and completely rigid cone center to provide pure piston action.



AUDIOM 755
15" woofer

AUDIOM 755: 15" woofer, voice coil diameter 3", fundamental resonance 25 cycles, power handling capacity 40 watts, total flux, 240,000 Maxwells.

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EDWARD TATNALL CANBY*

Who's Talkin' Stereo?

1. CHARLOTTE RUSSE IN HI-FI

Portraits in Sound. (Espana, The Moldau, Sorcerer's Apprentice, Russian Easter). Concert Arts Orch., Leinsdorf.

Capitol P 8446

Youngish Erich Leinsdorf has been one of the most brilliant recent conductors of Wagner, Mozart and other such, especially in opera; now he is going the way of the successful, moving on to the big-time semi-pops kind of classical recording, for one outfit after another. He did up all the Mozart symphonies for Westminster, but shifted quickly to these slightly overperformed items for the bigger outfit, Capitol, is now ripping out Italian opera for RCA Victor. If all goes as it should, he'll round out his days back in the classical fold again as a Maestro. (They call him that already at RCA; I heard it with my own ears.)

All of which is a prelude to the observation that this is just another of those slick hi-fi platters, and musically an unusually dull one for my taste. Just a job, and all the Leinsdorf reputation evidently hasn't had the slightest effect upon these already-slick players. It bangs and it sizzles expertly, this record, but the whole business is as cold as a fish. Fine for noisy background music.

Concert Russe. (Bald Mountain, Polovetsian Maidens, Marche Slave, Kamarinskaya). Pittsubrgh Symphony, Steinberg.

Capitol P 8450

When Capitol goes after an idea, it really pushes. Here's another of the same—but with a difference.

The Capitol sound is just as good, the music just as familiar, the program just as hi-fi. Steinberg, too, is one of our upcoming dynamic conductors. But maybe he hasn't been around as much as Leinsdorf—he hasn't learned how to do a slick job yet. He still plays this music as though it were exciting and interesting.

Also—Steinberg is conducting his own men here, and getting his own well-tried effects. It's his orchestra, technically a minor-league outfit trying for a big reputation, and it works hard for him. Thus though this record outwardly is a twin of the Leinsdorf "Portraits in Sound" and on your hi-fi set it'll come out very much the same, the music itself is 100 per cent better.

It can be done—you *can* have the fi and the music too. But it takes a dedicated, fine musician and an orchestra that follows him with the same sincerity. Steinberg has it. Leinsdorf doesn't.

Tchaikowsky - Borodin - Rimsky - Moussorgsky. . . . Cleveland Orch., Szell.

Epic LC 3483

This is Epic's contribution to the hi-fi chestnut department and is it a wow! Terrific.

The feature here is undoubtedly the sound, which is about the best I've ever heard in these works. Maybe I should say, it's the microphoning, plus high-quality processing of the usual and expected sort; the mike pickup here is one of these just-right miracles of calculation or good luck that bring music to life and make you think—why all the fuss over stereo? I had intended merely to sample this, expecting "more of the same" where I'd already had more than I could take—but I stayed with it to the end. Superb sound.

Szell, too, conducts his own orchestra, also technically a minor-league outfit that is growing fast, working hard to climb into the big-time category. Like the Pittsburgh, it does its work with spirit and attention. Like the Pittsburgh, it still is able, under a hard-working conductor, to play the chestnuts as though they were music. They are, oddly enough, though sometimes (especially at a High Fidelity Show) you'd never know it.

What music is played? I almost forgot. Epic didn't bother with a fancy (and short) subtitle and I don't have room to list all the familiar pieces. The only semi-surprise is "Dawn on the Moskva River" from Moussorgsky's "Khovantchina." You'll recognize it too, probably.

A Program of Russian Music. Royal Philharmonic, Efrem Kurtz.

Captol-EMI G7106

Here we go again—but with a different difference. Sorry, I simpy couldn't bear to hear that blasted "Capriccio Espagnol" of Rimsky-K. again—how many times can any one musical human being be expected to listen to it, what with all the other music in the world crying to be heard! But the rest of the program here is decidedly pleasing and the lively playing plus excellent sound will apply to Rimsky if you want to listen.

The name Kurtz may stir hi-fi recollections for you, of the first "hi-fi" LP recordings, out of Columbia with the N. Y. Philharmonic under this same man. Remember the Sword Dance from the "Gayne" Suite of Khatchaturian? If so, then you'll expect a vivid production here, as I did, and you'll be rewarded. The Royal Philharmonic fairly bubbles and the music, other than Rimsky, is imaginatively chosen for fresh hi-fi.

One whole side is given to Liadov, including not only his familiar "Enchanted Lake" and the little "Musical Snuff Box" that you've heard a million times, but two short tone poems in the manner of the "Sorcerer's Apprentice" (Russian-style) that are full of orchestral color. The other side matches the Capriccio with an old friend of mine, the Overture on Hebrew Themes of Prokofieff, in an orchestral transcription. This little piece, written in New York, in 1919, has always had for me a real New York flavor, a sort of Bronx cheer sound, that any Manhattan cab driver would recognize. To be sure, it's a bit watered down in this big British-played transcription (the original is for clarinet, piano, and string quartet) but some of the Manhattan taste remains.

Keep an eye peeled for more of this man, hi-fi Kurtz. He's good.

2. OLD AND VERY NEW

Boulez: Le Marteau Sans Maître. Instr. Ensemble, Marie-Thérèse Cahn, contralto, cond. by the composer.

Messiaen: Oiseaux Exotiques. Yvonne Loriod, pf., Orch. cond. Rudolf Albert.

Westminster XWN 18746

Boulez: Le Marteau Sans Maître.

Stockhausen: Nr. 5 Zeitmasse. Marjorie MacKay, contralto, instr. ensemble, Robt. Craft.

Columbia ML 5275

Time surely does march on, in music. Time was when "modern" music was Stravinsky's "Le Sacre" or even "Elektra" and "Salome" by Strauss; then, later, it was Gershwin, or Copland or Prokofieff or Hindemith. Now, you may want to know, modern music is Boulez, Stockhausen & Co.—and is it modern!

Well, actually not so modern if you mean, does it have antecedents, does it follow in a tradition. Definitely, these works do—the tradition, mainly, of Schoenberg and that archminiaturist of so many years ago, Anton Webern. If you've ever heard "Pierrot Lunaire" of Schoenberg, dating from 1912 if I remember rightly, you'll recognize this Boulez piece as its grandchild.

The modern music of this advanced and very influential school today is all twitters and squeaks and jingles. It has no rhythm that you can beat time to, no themes, no tunes (of the usual sort), no harmonies, no key—especially, no key. Tonality is so far removed from these composers that, at last, you can listen to them in perfect comfort, feeling that not so much as a suggestion of old-fashioned key will creep in. Good, if you ask me. (It's the modern pieces that fall into keys by accident, clumsily, that kill me.) The music is serial, a useful term for what has been called "twelve-tone" in the past. (The principle of serial repetition is larger and more general than the rigid pattern of twelve repeated tones.)

Two things will strike you as you listen here. One, the music is fabulously, incredibly complicated, enormously mathematical, straight out of the age of the computer, so complicated that you cannot believe it is seriously being played according to a repeatable and predictable pattern from a written score; and, two, it not only is played, from notes, but—if you will allow your ear to browse casually in it—the stuff is quite easy to listen to and rather pleasing, even on first impact.

Not pleasing if you are looking for tunes, or any other pre-conceived sort of sound. But if you will forget for the moment that this is music and think of it simply as organized, patterned sound-effect, then you'll find yourself amused, pleased, titillated, perhaps even fascinated. I was.

And if you want to prove that Boulez, at least, is practical and playable, here you have not one but two recordings! The Westminster is under the composer's direction and is the least lively of the two; the Columbia version, under the scholarly Robert Craft, is better recorded, livelier, more forceful—but maybe no more authentic; the point is, though, that you can

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

Fig. 3. View of complete console with top up and ready for use. Grille-cloth areas cover two AR-1 speaker systems.

ganged playback volume controls, and is a three-head machine—separate erase, record, and stacked playback heads. It will be noted that the recorder sits in a "well" in the cabinet (See *Fig.* 1). For recording "on location" it can be quickly disconnected without affecting the operation of the remaining components.

Another reason for selecting the Bogen preamp is the fact that it includes a tape monitor switch, which permits a program to be monitored from the tape as it is being recorded. None of the stereo preamps as yet announced seem to incorporate this feature, which is a most useful one.

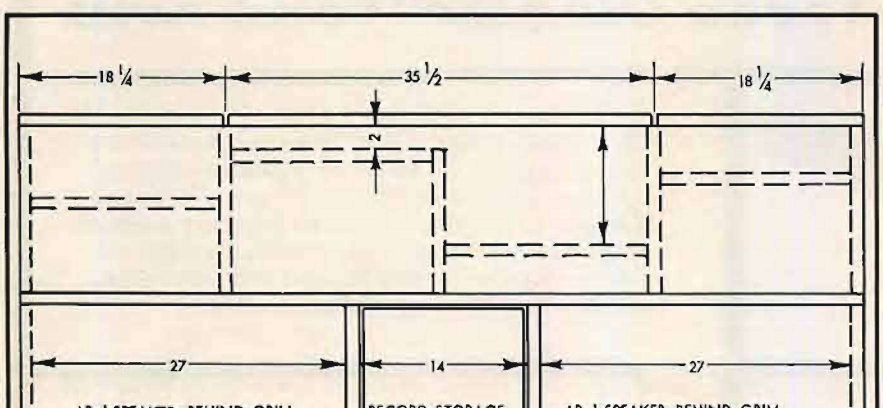
Operation

For monophonic operation, everything is straightforward—the main output of the preamp is paralleled to both power amplifiers and all preamp controls are fully effective on both channels. For stereo operation, however, it is necessary for the right channel from the tape recorder to be connected to the right power amplifier and speaker system, and accordingly one of the selector pushbuttons in the Bogen was converted to a transfer switch (This could be done externally by means of an added switch as shown in the block diagram, *Fig.* 2, if a preamp other than the Bogen is used). When the button is *not* depressed, the main output of the preamp is fed to both power amplifiers. Depressing the button, however, disconnects the right amplifier input from the main output of the preamp and transfers it to the right output of the tape recorder. The block diagram should make this clear. The preamp controls are now only operative on the left channel and the right channel has no controls whatever other than the volume control on the tape recorder.

Phasing Problems

So far so good, except that it was here we ran into our first snag. As will be appreciated, the right channel goes di-

(*Continued on page 80*)



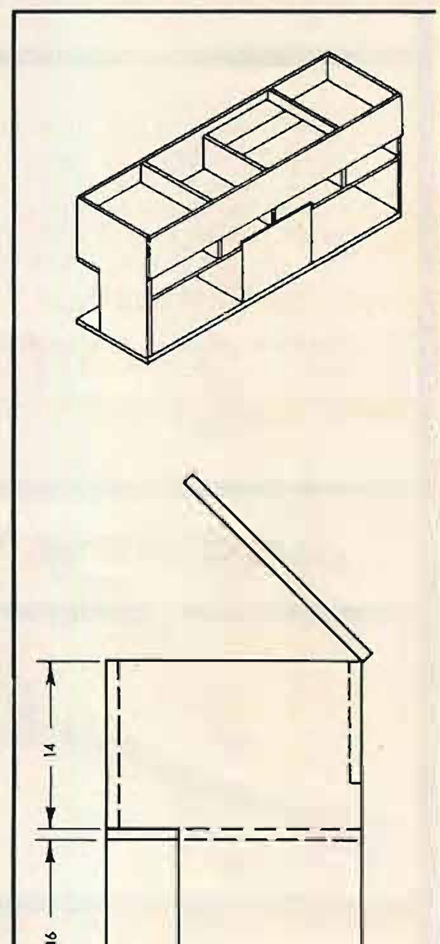


Fig. 4. Dimensional drawings of cabinet construction: left, front elevation; right, end and isometric view.

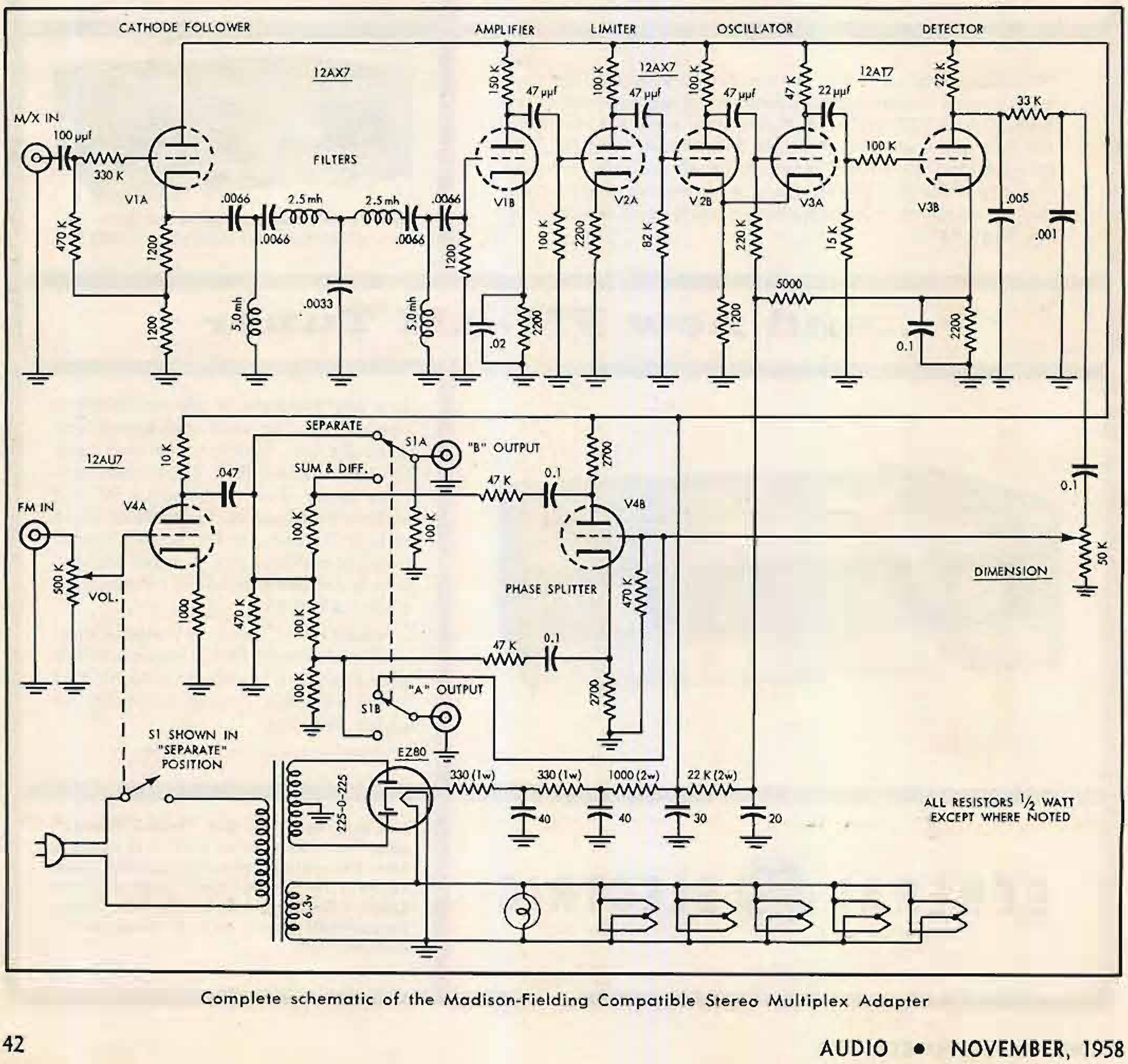
Compatible Stereo Multiplex Adapter – Complete Schematic

This schematic—published last month without values because of normal manufacturing secrecy—is repeated with all components identified now that the unit is on the market.

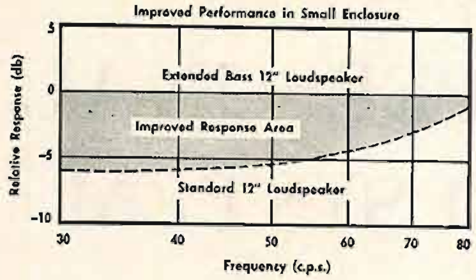
If there is any one item in the entire high-fidelity business that has recently captured the interest of experimenter and casual listener alike, it is the possibility of stereo broadcasting by FM multiplex. We have had a larger than usual response from readers regarding the Madison-Fielding multiplex adapter which was described in the October issue—some berating us because of publishing a schematic without values and others of more kindly persuasion asking that we might furnish the information to them individually.

Although all of the licensees of the Crosby Compatible Multiplex System have the information about the adapters, each manufacturer has his own ideas as to how the final unit should be built—both as to appearance and in final details of the circuit. For instance, the circuit of the M-F adapter differs slightly in the production models from the schematic shown in the October issue—one resistor has been added and one value changed.

We did not furnish the values with the original article at the request of the manufacturer. At our request he has consented to our running the schematic again with all values shown, and we trust this will answer the demands of our readers. Æ



Complete schematic of the Madison-Fielding Compatible Stereo Multiplex Adapter



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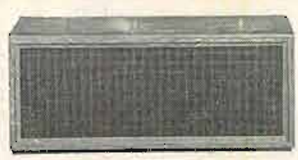

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FA-690 DUAL FM-AM STEREO TUNER PREAMPLIFIER – Embodies on one chassis two separate ultra-sensitive tuners for FM and AM, and a complete stereo preamplifier. The FM and AM tuners operate independently of each other for FM-AM stereo, and an FM-FM multiplex position is provided. Adjustable muting circuit eliminates interstation noise. Two illuminated tuning meters provide precise center-of-channel tuning for lowest distortion on FM, peak tuning for AM. The AM section features a broad/narrow band-width selector. The preamplifier section has two identical preamp units with ganged volume, equalization and tone controls plus a stereo balance control. Inputs provided for stereo records, stereo tape heads, microphones and tape amplifiers for auxiliary use. *$269.50.*
FA-680 DUAL FM-AM STEREO TUNER. Identical to the FA-690 except that it has no stereo preamplifier section. FA-680 *$199.50.* Both supplied with modern, low silhouette enclosures.

SM-245 STEREO PREAMP and CONTROL AMPLIFIER. Complete control system with self-contained dual channel preamp, featuring bass and treble controls, loudness and volume controls, and two power amplifiers rated at 16 watts each (32 watts peak each) at less than 1% distortion. Inputs are provided for stereo FM-AM broadcasts, stereo tape, stereo discs, microphones, and auxiliary, with a separate output for making stereo tape recordings. A balance control adjusts the relative level of the two channels. Illuminated slots on selector switch indicate function in use. Equally efficient for monaural application. Complete with enclosure, *$189.50.*

FA-670 FM-AM TUNER. The FM section features superior sensitivity, drift-free operation with a wide-band FM detector. Features a panel-mounted, illuminated tuning meter for precise center-of-channel tuning for low distortion; dual limiters, interstage muting, and high gain IF for reception of distant stations with virtually inaudible background noise; and an FM multiplex output for stereo broadcast reception. The AM tuner is a broadband superheterodyne type with a high-gain pentode RF amplifier, high gain IF stages, a 10KC interstation whistle filter, and a built in AM antenna. Has high inertia flywheel tuning, cathode follower low impedance output for use with long cable, an output level control. Complete *$179.50.*
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Both tuners complete with enclosure

SP-210 STEREO AMPLIFIER. Consists of two identical preamplifiers with ganged controls for balanced stereo operation. Premium type low-noise triodes are used in all low level stages. DC is supplied to all tube heaters for minimum hum. Inputs with equalization for all possible stereo source material—phonograph, tape, microphone, FM-AM, and FM multiplex. High sensitivity makes it ideal for magnetic stereo cartridges or tape heads. A balance control varies the signal to each speaker for best stereo performance. Auxiliary outputs for making stereo tape recordings. Complete with enclosure, *$89.50.*

SP-216 STEREO PREAMP and AUDIO CONTROL. Professional-type, dual-channel preamplifier with inputs for stereo FM-AM broadcasts, stereo tapes, stereo discs, microphones, and other stereo signal sources. Has separate output for recording stereo tapes. Two VU meters and controls for setting reference and peak levels and separate output for recording stereo tapes makes this unit ideal for recording. Monitor/Record switch on front panel indicates record output level or the relative level of the signal at audio output jacks for balancing both channels in stereo. Automatic shut-off position on power switch turns off entire system after last record is played. Features bass, treble, volume and loudness controls, and balance control for equalizing level between both channels. May be used with the SA-232 or SA-260 stereo power amplifiers. Complete with enclosure, *$189.50.*

SA-232 AND SA-260 BASIC STEREO AMPLIFIERS. The SA-232 and SA-260 Basic Stereo Amplifiers each consisting of two identical power amplifiers, incorporate the latest advances in the art of high fidelity negative feedback amplifier design. The SA-232 delivers a total of 32 watts of undistorted power (64 watts peak), the maximum power obtainable without exceeding the tube manufacturer's specifications. The SA-260, with 60 watts of undistorted power (120 watts peak) is well within operating characteristics of its output tubes. Power tap-offs for operating the SP-210 Stereo Preamplifier.
Both units are supplied with brass finished metal covers. SA-232, *$89.50.* SA-260, *$129.50.*

These identical stereophonic components are also incorporated in Pilot custom component-consoles.

Complete individual specifications available from Pilot on request.

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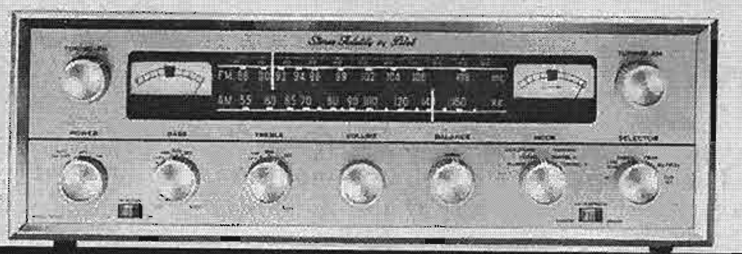
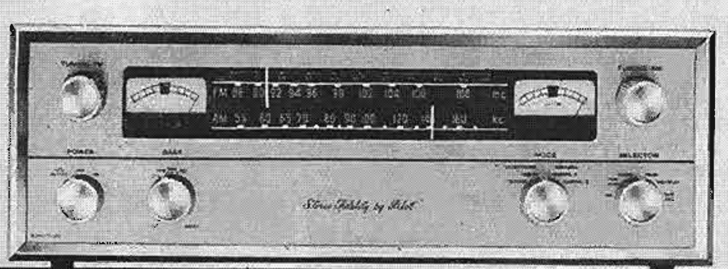


Pilot FA-690, Deluxe Stereo FM-AM Tuner and Preamplifier, **$269.50**



Pilot FA-680, Deluxe Stereo FM-AM Tuner, **$199.50**

Pilot SM-245, Stereo Preamp and Control Amplifier, **$189.50**

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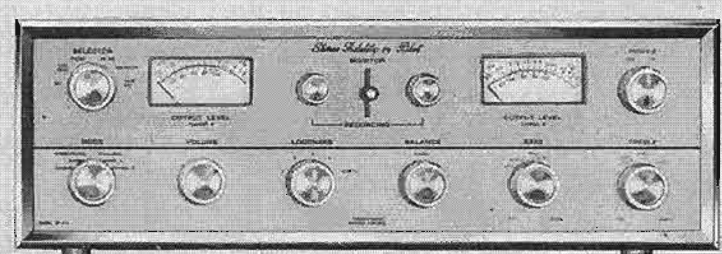
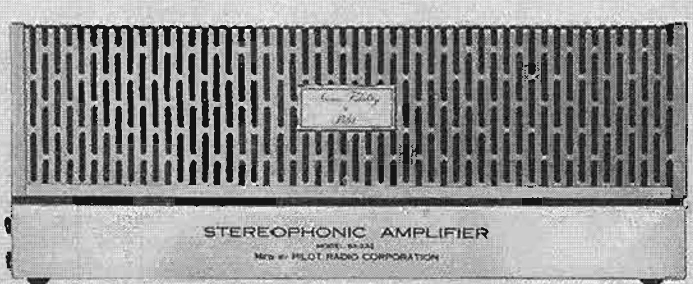
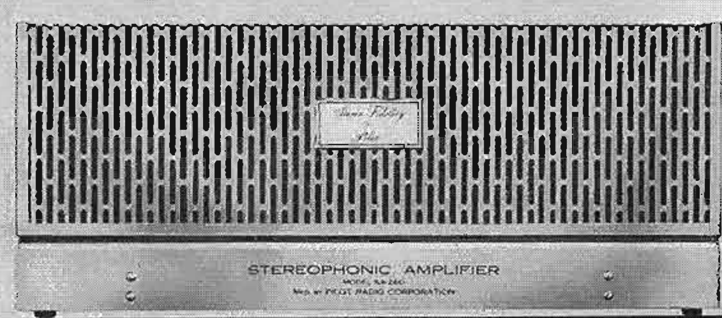


Pilot SP-216, Stereo Preamp and Audio Control, **$189.50**



Pilot SA-232 (64 Watts Peak), Stereo Basic Amplifier, **$89.50**



Pilot SA-260 (120 Watts Peak), Stereo Basic Amplifier, **$129.50**

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Equipment Review

Harman-Kardon "Trio" Tri-plex Stereophonic Amplifier —United Speaker Systems' Models X-100 and X-100-E

CONVERSION TO STEREO has become the principal problem to many audiofans —mainly because they are not yet completely familiar with what facilities they are likely to require. Either they have a complete monophonic system already and feel that from an economic standpoint they must use as much of it as possible, or else they plan to start from scratch, adding from time to time as they become more familiar with their needs.

The Harman-Kardon "Trio," Model A-224, serves—as its name indicates—in three roles. First, it is a complete stereo amplifier system with 12 watts of audio in each channel; second, it is a monophonic amplifier with 24 watts of audio (leaving the preamplifiers and tone-control stages unused); and third, it is a conversion amplifier which can serve as the input section of a stereophonic system and one of the two output amplifiers—one which provides 24 watts of power—with the other being one already in the user's monophonic system.

In its first role—that of a complete stereophonic amplifier—the Trio provides two equalized preamplifier stages, usable for either phono or tape head inputs, and also accommodates three high-level inputs, such as tuner, tape preamp output, or high-output phono pickup, such as a ceramic. In the latter inputs, the impedance offered to the cartridge is 2 megohms, which is sufficient for good low-frequency response with a ceramic cartridge. The preamplifiers are 12AX7's, with feedback equalization, switchable on the rear panel of the amplifier to phono or tape positions. These are followed by one section of a 12AU7 in each channel to provide sufficient gain for the tone controls, and these are followed in turn by 12AX7's as gain and phase-splitter stages, which drive a pair of EL84's in each output channel.

Tone controls are ganged for the two channels, with bass and treble being separate as in all high-fidelity equipment. The volume-loudness control is made to serve in the chosen role by means of a slide switch, and another slide switch serves to control the rumble filter. The MODE switch provides for stereo, stereo reverse, and either right or left input channel to both outputs as a monophonic amplifier. The FUNCTION control selects the input source.

Two other slide switches provide an unusual form of control which would be ideal in many installations. One switch, with positions labeled ONE and ALL, controls whether one or two pairs of output lines are being fed, and the other controls which of the two output pairs is being fed when the first switch is set at ONE. Thus the user can feed one speaker system in his living room or another in his den, or he may feed both at the same time if he wishes. This provides a sort of flexibility in speaker control that is rarely found on an amplifier control panel. Similar control action works as well when the amplifier is used in the monophonic application.

For both monophonic and stereo-conversion applications, it is necessary to switch the operation from SEPARATE to PARALLEL. This is done by means of a slide switch located inside the amplifier on the apron behind the front panel. This switch connects both output amplifiers to the right input section, and feeds the left input section to an output jack located on the rear panel of the amplifier. When this is done, it is necessary to strap the speaker connections of both output sections together. Both amplifiers have output impedances of 8, 16, and 32 ohms, and the instructions indicate that when a 16-ohm speaker is used in the monophonic or stereo-conversion application, both 32-ohm terminals should be strapped together to feed a 16 ohm speaker and so on. In the usual manner, the second (left) channel is fed to the input of a second power amplifier and thence to its own speaker system. The speaker selector switches are inoperative in the stereo-conversion connection, but they still function in the monophonic use.

Sensitivity of the amplifier is such that a 3-mv signal at the phono input or a 1-mv signal at the tape input will give the rated 12 watt output, and a similar output is had from a 300-mv signal at the AUX, TUNER, and high-level phono inputs. A tape output is provided ahead of both tone and volume controls, with a 1-volt signal for the indicated inputs. The output for the left-channel external amplifier is 0.5 volts. Tone control ranges are ±12 db at 50 and 10,000 cps, with the rumble filter giving a rolloff of 12 db per octave below 50 cps.

The Trio may be used in its normal metal housing (at extra cost) or it may be mounted behind a conventional panel without the enclosure. In operation, it was found to be convenient, and to have adequate control flexibility for practically any installation. The only change we might suggest in the entire unit is that it might provide both tape and phono input jacks for the preamplifiers, and to have the FUNCTION control switch inputs from tape to phono (and change the equalization at the same time) as a front panel control, rather than making it necessary to change input plugs and throw the switch on the rear panel. This is an amplifier we could recommend heartily for any one going through the pangs of conversion from mono to stereo. **M-27**

Fig. 1. The Harman-Kardon "Trio," Model A-224—a multipurpose stereo-monophonic-conversion amplifier.

UNITED SPEAKER SYSTEMS' MODEL X-100

Back in September, 1956, after comparing the United "Premiere" speaker system with our standard system, we wrote, "On the whole, however, we consider this to be one of the very few that we have heard that compares so closely to the comparison speaker." The old Premiere, with its 800-cps crossover and high-frequency horn system, has been improved—even though the improvement is only noticeable when direct comparison is made with the older model—by the change to a 500-cps crossover and the necessary change to a larger horn. The improvement is not glaring by a long ways, but after careful listening is must be admitted that there is a slight improvement in the midrange with the Premiere 500. All of this is beside the point, however, since we are primarily discussing the new and considerably less expensive X-100, shown in *Fig.* 2. This model shows the importance of good enclosure design in getting good performance out of medium-priced components, for it is perfectly obvious that an Altec 803A woofer and an Altec 802A driver with the necessary high-frequency horn can not be furnished at the price of the X-100.

However, we are inclined to say that—like much high-fidelity equipment—the differences in performance are rarely noticed *on average program material* during a typical listening session. Of course there is a difference between a $300 ultra-super-special FM tuner and the economy model at $19.95 (if there is such). But on a high percentage of broadcast material the difference might not be noticed. In the same way, there is a difference between the X-100 and the Premiere, but on 75 per cent of the program material the difference might not be noticed.

Using our customary test record for low-frequency performance, the X-100 was observed to be a good performer from about 37 cps up, with the upper limit beyond our own hearing range. The response is smooth throughout the entire range, and the coloration—if there is any—is similar to that of the Premiere, which in turn is similar to our Standard system. On a subjective basis, which is the only way we feel that speaker testing can be done without anechoic chambers and fabulously expensive measuring equipment (and who listens to speakers in an anechoic chamber anyway?), we believe the X-100 to be an

(*Continued on page 79*)

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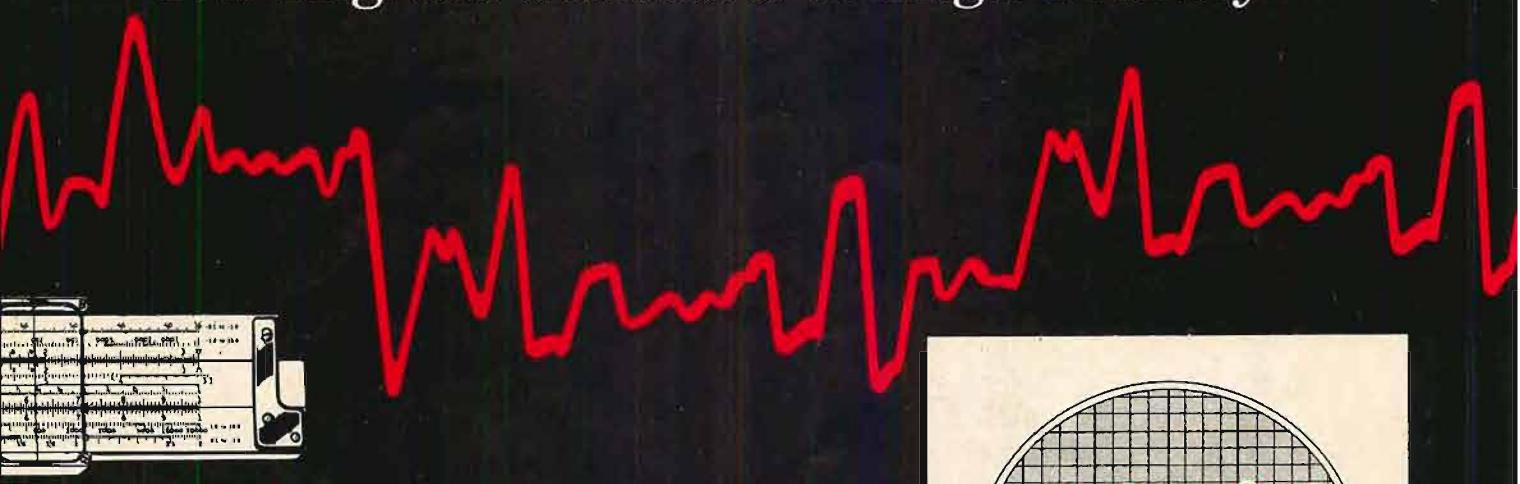

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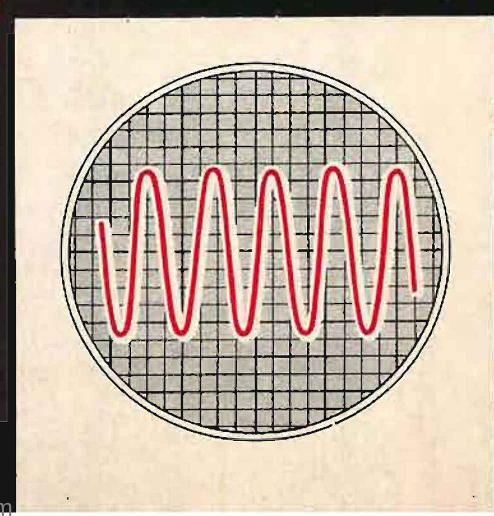

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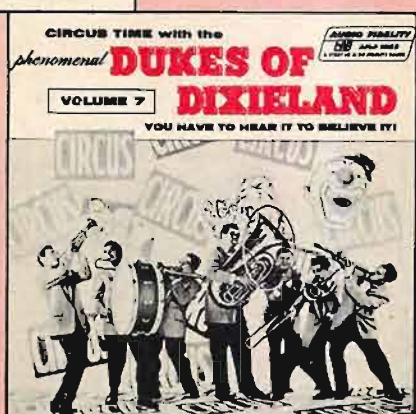

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actually "recognize" the same music, if you make a direct AB comparison. Amazing.

Stockhausen is the German equivalent of Boulez, sounding superficially like him—clearly of the same school of thought and the same time in history. But you'll be gratified, perhaps, to detect certain typical and traditional Germanisms in him—a more emotional, high-tension expression, an urge to "say something," something big, as all good Germans seem to have to do, an almost contrapuntal structure, distantly out of old Bach and Beethoven.

As for Messiaen, this somewhat older French composer (born 1908) seems to have heaved himself out of a rut of rather narrowly religious music of unpleasant yet old-fashioned dissonance, straight into the new school—or as near to it as he can manage. His "Strange Birds" are strange indeed, for piano and orchestra, as complex and as fragmentary as the work of the other two but somehow less modern in sound, with even a trace now and then of theme and harmony. Oddly enough, this music is actually about birds, some forty of them, and it is full of Greek and Hindu rhythms, as well . . . but if you start reading the technical notes on the record you'll lose your nerve. Don't. Just listen.

Debussy: La Mer. Ravel: Mother Goose Suite. L'Orch. de la Suisse Romande, Ansermet. **Richmond B 19007**

A new $1.98 label and by all the evidence a top-notch one, for these Richmonds are no less than those dignified old examples of the famed London ffrr recording in the days when ffrr was the most advanced sort of sound we had. Reissues. The new records are marked "made in U.S.A." which implies that they're pressed here, where most original Londons (maybe all?) have been pressed in England.

This one is a crackerjack and I'll have to assume that it is typical. Very sympathetic performances of both works (the Ravel work has the Introduction, often omitted) and the sound is, as we used to say, "typical ffrr." In comparison with the very latest I'd say it has a few slight whiskers, a bit of extra brilliance, particularly in the strings, that maybe wasn't in the actual playing. But this sound, remember, was the sort that we thought of as unsurpassable only a short while ago. The latest and best sound is simply a trifle more clean-shaven, by an hour or two at most. Nicely quiet pressing on these, too. You'll find a couple of dozen already on the market, but we won't be able to cover them all here.

Florent Schmitt: Une Semaine du Petit Ferme-L'Oeil; Trois Rapsodies (two pianos). Gaby and Robert Casadesus, duo-pianists. **Columbia ML 5259**

Florent Schmitt is a Frenchman's Frenchman, musically speaking—a composer much revered in France and not much known elsewhere. His two-piano music, however, is glib and easy to take for our ears and is marvelously written for the medium.

Schmitt is (last I knew) still alive at 88, but his characteristic musical output seems to have been in the pre-War I period, along with Debussy and Ravel. These two record sides date from 1912 and 1903 respectively, the first a very French nursery-style suite (cf. Ravel's "Mother Goose" or Debussy's "Children's Corner") complete with titles—the story is Hans Christian Andersen— and the second a group of three big virtuoso pieces in a semi-nationalistic idiom, French, Polish, Viennese, all sounding thoroughly French as we hear them now. The two Casadesus pianists do a noble job in virtuoso playing though, as always, their playing is on the muscular side. But this is their own special music, nationally speaking, and they lavish unusual care on its expression.

Dvorak: String Quartets in E Flat, Op. 51, in F, Op. 96 ("American"). Netherlands Quartet. **Epic LC 3490**

The jolly, songful, Romantic Dvorak quartets make an easy introduction to chamber music for anybody who doesn't expect to find it an easy sort of listening. The "American" quartet was composed out in Spillville, Iowa, in the early Nineties—Dvorak was teaching in New York and went to a Czech settlement there for a quiet working vacation. The earlier quartet is all Bohemia, full of lilting tunes and dirndl-type rhythms.

Incidentally, it says here that Dvorak was nuts over trains and would drive (horse and buggy) for an hour or so up to 155th St. in New York to hear the "Chicago Express" go by—whatever that might be. Maybe the Central's pre-Twentieth Century flyer? He had to be hauled away from the train yards, just as old Walt Whitman rode the N. Y. trolley cars. So, *quid pro quo* (tit for tat) if you'll listen to Dvorak's music, you can guess that he might enjoy listening to one of our hi-fi rail dynamic discs, were he around. (He'd be a ripe 117 years old, right now.)

Mozart: Piano Sonata, Four Hands, in F, K. 497; Andante with Five Vars., K. 501; Adagio and Allegro in F Minor (mechanical organ), **K. 594.** Lilly Berger, Fritz Neumayer, Mozart piano. **Archive ARC 3101**

The Archive series from Decca (Deutsche Grammophon) is a never ending pleasure for me even when, as here, the performance is stodgy. That can happen in Germany or Austria—but so can the brilliant planning that leads to this enterprising and virtually limitless survey of older music in its own terms.

Here, once more, is the fascinating Mozart piano, the instrument of the 1780's upon which Mozart played and for which he wrote his music. This one is real, built in 1780 eleven years before Mozart died and restored to usefulness in the mid-thirties of this century—the period when it suddenly struck a lot of poeple that these old instruments might be worth listening to. This record is unique in that four hands do the playing in each of the works.

Mozart wrote a considerable amount of piano-four-hands music, mainly for himself and various pupils to play together. Among these works are some of his finest though their utilitarian function would not have suggested such a possibility.

Yet, until now, virtually nobody alive had any idea as to the actual sound that might have come forth when Mozart himself and his pupil (almost invariably feminine) sat down on the same piano bench and went to work. Here you have it, at least in the outward, physical aspect, and it is really lovely, after the first five minutes of adjusting to an unexpected and more metallic range of tone color.

True—the Mozart instrument was technically in an early stage of piano development. It was far removed from the later Steinways. But, as you might expect, Mozart's own music sounds really a lot better, more balanced, sharper and more clear, on this instrument than on relatively muffled and heavy-bottomed modern piano. The sound is a cross between a harpsichord and a piano, more percussive by far than our piano, more edgy, but with a surprising range of bright tone colors from the loud, banging *Forte* to the almost flute-like soft tones. A twangy bass, a bright treble, a clarinet-like middle range, all give lift and lightness to music that sounds too often thin-blooded and precious on our own pianos. Not on this one.

Unfortunately, as I say, the performance by these two careful pianists is almost as pedantic as the piano itself is vibrantly alive. They might as well be sight reading to a metronome, for all the expression or understanding they show. Phrasing and tone-production are good; the playing is expert enough. But unimaginative is the best I can say for the general effect.

Vincent Lübeck; Nicolaus Bruhns: Organ Music. Hans Heintze, Organ of St. Johannis, Lüneberg. **Archive ARC 3094**

Pursuing systematically its coverage of all the great music of the past and in particular German music, the Archive series here produces two of the leaders in what used to be called, ever so slightingly, the "pre-Bach" school of organ composers in North Germany. They were far more important in themselves than the term would suggest, as you'll hear on this record.

Lübeck, whose name I'd heard and some of his scant music too, is the least interesting as presented in this performance. Three preludes and fugues and a Partita on a typical German chorale tune are sprightly enough, yet somehow a bit dull in the harmonies, somewhat on the didactic side. An expert composer in the then flourishing organ style, but perhaps not a man of top imagination.

The other one, Bruhns, is a name few of us will ever have heard and it's the more surprising to find that this short-lived composer was one of the really great dramatic organists of his time. You'll notice the difference right away.

Nice playing on a fine organ, originally installed in the mid-fifteen-hundreds and polished off about 1912. It is a mere 400 years old now and it sounds better than ever thanks to a valve and cylinder job and general overhaul back around 1952.

(If you think "antique" cars in perfect running order are something to get excited about today, just wait until they get restored at age 400! It's not surprising that organists and music listeners get enthused about these instruments.)

P.S. Bruhns must be a typical North German name. Note the similarity to Brahms, who came from Hamburg, not far from Bruhns' locality.

None But the Lonely Heart (Jennie Tourel sings Russian Love Songs). Brooks Smith, pf. **Decca DL 9981**

Jennie Tourel, late of Columbia Records, is one of our great dramatic singers and a natural-born mike artist. She isn't projected here as well as she might be, on her first Decca disc.

The trouble—if it is trouble—would seem to be the mike set-up, which puts the lady in a somewhat off-mike position and cuts down on the persuasive powers of her very vocal Russian. (All the songs are sung in Russian.) It's a matter of taste and artistic judgment, of course, and Decca's recording people may simply think differently than Columbia's. Yes, I know that the Tourel voice is powerful and not too easy to mike at close range. Still—I think she'd be much better if she had a more direct presence here and I'm sure that modern recording circuits could take it, too.

It just might be that Decca is out of practice in classical recording technique. The company has for a long time followed a policy of classical importing, with only a few sparse domestic classical recordings to keep the ball rolling. With most other outfits working year in and year out like mad, it's reasonable to suppose that Decca might lose a bit of that tricky know-how that must go into every recording, of any sort. Just a thought—maybe Decca just plain wanted it like this.

Erika Koeth Sings Arias of Mozart, Richard Strauss. Berlin Philharmonic, Schuchter; Matzereth. **Capitol-EMI G 7114**

So sorry, but I don't quite go along with the enthusiastic quotes on the record jacket concerning this lady, who is lauded for the "perfection, intensity and dramatic force that make her the foremost dramatic soprano on the continent"—at least, not on this record. Sure, every soprano is the greatest when the quotes start rolling, and maybe this one is terrific in the flesh and on the stage. But I find her an authentic Mozart-Strauss soprano (that is, she knows the style of singing, she has the proper tone color and vocal production in the Germanic manner) but she just doesn't give very much. A second-rate singer of the top rank, if you see what I mean. Next to the really best, she comes off second-best.

(Watch these columns in future—maybe her next recording will receive raves from me, all unwittingly!)

Bach: Mass in B Minor. Chorus, Symphony of the Bavarian Radio, Lois Marshall, Hertha Hopper, Peter Pears, Kim Borg, Hans Braun; cond. Eugen Jochum. **Epic SC 6027**

I hadn't listened to the good old Bach Mass, eternal favorite of the amateur choral singer, for I don't know how many years when I hit this album. (Last time, I guess, was the Westminster album with Scherchen, for which I wrote liner notes.) I was knocked for a loop by this performance and I'm not entirely sure how much was old man Bach himself and how much Herr Eugen Jochum and his very willing cohorts.

Anyhow, this version is beautifully recorded, sung and played with utter sincerity and aliveness, yet done without more than a very few eccentricities, of the sort that too many big-name conductors feel they must perpetuate. The music just rolls along, and all of us who have actually sung in the Mass will be enchanted—so will anybody who has even an inkling of a taste for the big piece. Even the solos are excellent, with one exception (contralto)—and she's merely so-so.

Note the unexpected appearance of the British tenor, Peter Pears, and the definitely non-German Lois Marshall, in the internationally slanted solo group. Kim Borg, in case you wondered, is the basso profundo and no relation to Kim Novak.

Mozart: Requiem, K. 626. Soloists, Choir of St. Hedwig's, Berlin Philharmonic, Kempe. **Capitol-EMI G 7113**

I'm in the midst of this record as final deadline approaches, but must quickly recommend it even without having reached the end. Anything from the youthful St. Hedwig's Choir in Berlin is worth your immediate attention if you love big choral works; these singers do a splendid job, exciting, disciplined, with grandeur yet a marvelous leanness of tone and accuracy of pitch for such a large chorus. No wobbles, no heaviness. You'll find them in other works, too, notably the Brahms Requiem (twice).

The only immediate rival to this splendid recording that comes to mind is the recent version with Beecham (Columbia ML 5160), a very British version, as this is characteristically and energetically Berlinese. (There are, to be sure, numerous other versions available—I can't keep them all in my somewhat porous mind.) The Beecham, though, is more a virtuoso job, better in the solo voices, a bit stunty in a striking and legitimate way; this one is all meat and drink, the essence of the big piece, alive but uneccentric, in spite of so-so soloists. Most Mozart lovers will enjoy it on this score, and so will many a newcomer to this astonishing music from Mozart's deathbed.

Biggest mystery in art—the piece was completed by a pupil, Süssmeyer, a conscientious nobody who, somehow, absorbed Mozart's influence and personality to an almost hypnotic extent and was able after Mozart's death not only to round out unfinished sections of the manuscript but actually to compose additional numbers that simply cannot be told from the genuine Mozart, even by those who worship Mozart as the greatest creative genius in modern musical history. A strange thing.

Dowland: Ayres for Four Voices, Vol. 1. Marenzio, Monteverdi: Madrigals on Texts from "Il Pastor Fido." The Golden Age Singers.
Westminster XWN 18711, 18712

Curiosity led me to these two Westminster reissues, since I had enthused over one of them in the original 1955 release (AUDIO, Feb. 1956) but had missed the other. The new versions are simply re-packagings, out of the old and too-fancy-sealed WN series into the lower-priced XWN category, with new and attractive covers, the same notes, a newly quiet pressing (from the same disc masters as before). This Westminster reissue policy, combining repackaging with frequent re-cutting, or re-shuffling of musical combinations into various alternative formats, is all to the good for most record buyers. Only complaint: I get mixed up between the new and the old since the company isn't telling us which are reissues and which brand new recordings. Doesn't really matter, does it?

The Golden Age group is perhaps the best madrigal singing group now on records, well ahead of any other I can think of—far ahead of our American counterparts. They make the clearest harmony, they speak the best and most natural (British) diction, they make the most of phrasing and rhythm.

The first disc is all-Dowland, the music not actually madrigals but simply the four-part settings of Dowland's sweet, ear-catching songs; they also occur in solo-style form with instrumental accompaniment, originally done mostly on the lute, the then equivalent of our guitar. If you recognize "Come, come again, sweet love doth now invite," the first item on this record, you'll know the style of the music. An occasional lute joins in here for variety—it would have been better to sing every third or fourth piece as a solo with the same lute.

The other disc presents Italian madrigals on a group of then-popular texts, one setting by the earlier master, Marenzio, another by the later genius, Monteverdi. Utterly different treatment and the printed commentary here will help you study the very interesting comparison, word for word, as the two composers get down to business on the same texts.

3. SPECIALTIES

Schubert: Piano Trio #2 in E Flat, Op. 100. The Immaculate Heart Trio.
Capitol P 8442

I love Schubert—but, frankly, I got hold of this one for curiosity, as many a record buyer will do; I wanted to hear what three nuns would do with a lot of red-blooded Schubert.

They do fine—or at least two of the three do fine. The sisters are actual sisters, as well as being nuns with those nice masculine names they take on—they are Sisters Mark, Anthony and Denis. Anthony is the youngest and an excellent cellist, who played in symphony orchestra and movie work before she joined up; all of the three are highly trained musicians and music teachers. The pianist, Sister Mark, is imaginative and powerful as well as top-notch on her technique, as any Schubert pianist must be. Only the violin, Sister Denis, seems to me to be a follower rather than a leader; she just plays. You can't do that with Schubert, and so the trio is slightly weak in the upper department, which is a shame considering how nicely it plays as a group and how well Schubert is served by the other two.

Norwegian Songs by Arne Dörumsgaard. Kirsten Flagstad; Gerald Moore, pf.
Angel 35573

These classic, traditional "lieder," sung by the redoubtable Flagstad and played by a very great accompanist, Gerald Moore, sound as though perhaps they were composed during the middle years of the last century, or just possibly at the end, by a conservative musician. Actually, their author is a youngish man (well, he's younger than I am) from Norway, resident in Paris, with whom I spent an interesting evening a couple of years ago. He told me at that time that the great Flagstad often sang his music.

What is odd here, you see, is that a man can compose expertly in a style that is more or less of a long-past era. Not exactly—chances are that a publisher of 1850 would find some mildly revolutionary bits of harmony here and there according to 1850 thinking. But the music is not only un-modern in the usual sense of key and dissonance—it is of a type that no longer really exists, the art songs of Schubert, Schumann, Wolf, Brahms and the like.

And yet Dörumsgaard is good, and Flagstad is right. His music is expertly made, expressive, versatile; above all, it is highly musical! But it's old fashioned beyond belief.

The explanation is simple, I'd say. After all, we today have the odd habit of listening mainly to music of earlier times—to the point where its language is the musical sound we know best. It's not surprising, then, that those of us who come to the point of "speaking" music, composing it, sometimes speak in the very terms we have come to know, the general sound of much earlier music. Some of us (myself included) do not find it possible to say anything worth saying in that earlier language—it seems wrong, pointless, as though a writer of plays could write only in Shakespearian-style language.

Others, Dörumsgaard included, are not bothered by these thoughts and are able to turn out music by the ream—and have it played, sold, appreciated. Why not?

Virtuoso Guitar. Rey de la Torre.
Epic LC 3479

No flamenco here—this is classical guitar, and only a couple of items have a Spanish cast to their sound. Side 1 is a sonata by one Giuliani, from Vienna out of Italy, in a simple 18th century style that is most like the little piano Sonatinas of Beethoven, nothing much, but interesting as an example of what the guitar could and can do. The other record side ranges from Villa Lobos to a tremulo chestnut of virtuosity by Tarrega—every guitarist plays that one.

For my ear, the only stuff that has juice in it on this classical disc is the de Falla, comprising two items. Thank the Lord for him; he puts guts into Spanish music.

Paul Ulanowsky Accompanies You in Lieder Favorites. (Schumann, Schubert, Brahms). **Boston B-502 (10")**

This is an oldish item but I managed to miss it on its first appearance and it's still in the catalogue. An interesting do-it-yourself disc, with one of the two or three finest accompanists for the German lied now alive, playing the piano parts as though Lotte Lehmann or Irmgard Seefried or Suzanne Danco were singing the songs—he has been their partner at the piano on many an occasion.

It is astonishing how firmly these piano parts—without the song—stand up by themselves as interesting music when played by a master such as Ulanowsky. Every music appreciation teacher explains that the German lied, like the sonata for violin and piano, is an equal partnership between solo and "accompaniment," the piano perhaps ever more important than the solo itself. Here you have the direct evidence. Also, it's interesting to note (at least for those who think music is a mathematical affair) how irregular in terms of literal time values a good accompaniment can be. The notes technically may be of equal value, but nobody—nobody who has any music in him—plays them that way. Not Ulanowsky, anyhow, and he should know.

Companion to TV (from the original Sound Track of the Urban-Eclipse Silent Film, The Fatal Love).
Bu naB #5 (12" LP, all speeds.)

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....

(Well, what more do you expect me to say? You can find out from Orville K. Snav & Associates, 111 N. Jefferson, Mason City, Iowa. It's your risk.)

Folk Dances from Erin. The Dublin Quintet. **Westminster WF 12011**

This is definitely folk dance for dancing. I say so for the best of reasons; as a listening disc, it is enough to drive you crazy with monotony, and that is precisely as it should be.

Some people think that dance music of the kind that repeats over and over, like a Virginia Reel, should be tricked up so that on each repeat there is a difference of some sort—change the instrument, jump to a different key, add fancy variations. Yes, if for passive listening, armchair style.

But not for dancing! The essence of a good English or Irish folk dance and of many another too (including plenty of square dance music) is that the music repeats, but the *dance itself changes*. The variety and interest is in the dancing—new figures each time around, or new people dancing the same figures, all against a constant and steady musical backdrop.

So—this music starts and just goes on and on, each item lasting four or five minutes without a break, repeating the same music over and over. Authentic, proper, perfect for dancing. Or, for that matter, for background music to any old *activity* just so long as it has its own interest and variety, even to sweeping the floor or sawing wood and hammering nails or paying bills. But for goodness' sake, don't just sit and stare fixedly at your loudspeaker while this plays. Æ

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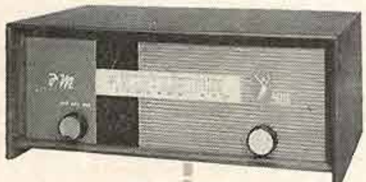

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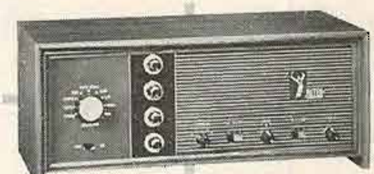

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JAZZ and all that

CHARLES A. ROBERTSON*

STEREO

Matty Matlock: The Dixieland Story
Warner Bros. BS1202
Sousa In Stereo Warner Bros. BS1209

When a record company acquired a motion picture studio with some of its profits a couple of years ago, other film firms looked at their pocketbooks and began casting eyes at that corner of the entertainment industry. First to take a step was Paramount, and the advent of stereo provided a good opening for Warner Bros. and United Artists to enter the fold. Both have their sights set on the popular market, but may fire a few divergent volleys. United Artists taped part of the Great South Bay Jazz Festival in stereo, and the staff at Warner Bros. now includes George Avakian, producer of many estimable jazz dates at Columbia, and more recently at World-Pacific.

Among the first releases on the Warner Bros. label, all issued simultaneously in stereo and monophonic versions, is a sumptuous two-disc collection of twenty-three Dixieland tunes. They range from numbers introduced by the Original Dixieland Band to *South Rampart St. Parade*, and Joe Sullivan's *Little Rock Getaway*. The chief anthologist is Matty Matlock, who chooses to back his liquid clarinet with pianist Stan Wrightsman; Nick Fatool, drums; George Van Eps, guitar; Morty Corb, bass; John Best and Shorty Sherock, trumpets; and trombonists Moe Schneider and Abe Lincoln.

Eddie Miller, another standby of the old Bob Crosby organization, appears on baritone sax and is largely responsible for the unique sound. In stereo, he is on the right to balance the clarinet on the left, with the drums nicely centered. Playing in a relaxed adaptation of the Crosby style, the group has enough power to duplicate the big band and the soloists to recreate the smaller Bobcat unit. Producer Lou Busch and the engineers concentrate on the music and there are no distracting effects.

The efforts of film studio engineers, up to now, were recognized on records mainly by way of various sound tracks. With stereo in the home, their long experience will be used more widely. As Warner Bros. uses its own studios on the West Coast, perhaps the first examples of their techniques are the ear-filling array of Sousa marches played by the Warner Bros. Military Band. The sound is big and bright, with all the immediacy of a newsreel. It is high-decibel stereo, especially when a male chorus enters on *Field Artillery March*, spread over a considerable area by a glockenspiel and a complement of drums. Henry Mancini conducts a dozen of Sousa's best, including the seldom-heard *National Fencible, Invincible Eagle*, and *Hands Across the Sea.*

Great Song Hits Of The Glenn Miller Orchestra Grand Award GA207 SD

At the head of the list of the ten albums in Grand Award's first stereo release is this recreation of a dozen hits from the legacy of Glenn Miller. Eighteen men fill the chairs in the All Star Alumni Orchestra, with the burdens of leader falling lightly on the shoulders of Bobby Byrne, a ringer from the Dorsey Bros. school. He sticks to his trombone, keeping a relaxed eye on the proceedings, and the band responds in kind. The original arrangements are followed, but Messrs. Beneke, Klink Abato, McMickle, Hackett, Hucko, and Al Mastren seem able to play from memory. It is the sort of performance rarely heard in a studio or elsewhere.

Lou Stein is back for the piano choruses on *String of Pearls, Moonlight Serenade*, and *Rhapsody In Blue*. Carmen Mastren, on guitar, makes one of his few jazz appearances to rejoin bassist Trigger Alpert and drummer Maurice Purtill. They are strongly propulsive and this is hardly an item for background listening. It calls for a big room and dancing feet. Engineer Bob Fine succeeds in obtaining a high level, matched on few monophonic records, by close miking. It will make small packaged systems sound twice as big, and is accomplished with no loss in playing time. Most Grand Award sides run well in excess of twenty minutes. Flexible controls on larger systems should be able to work it into shape without sacrificing too much brilliance, and the soloists are well-centered between the sections.

"Hawaiian Hits (GA208 SD)," finds Paul Whiteman balancing the full sweep of his string section against a romantic island guitar. The songs are arranged to meet Broadway standards, but Whiteman knows them all, from *My Little Grass Shack* to *Drifting and Dreaming*, having played them for several dancing generations.

One of the most entertaining uses of the stereo effect is demonstrated by Knuckles O'Toole playing "Honky Tonk Piano (GA204 SD)". The interplay of piano, banjo and drums can be realized only in stereo. It should be heard regardless of your feelings about this kind of piano, as it gives a rough idea of how a string trio will sound in stereo a year or so from now. When that time comes, you will be glad O'Toole was around to help pave the way. If you like an uninhibited pianist, a sterling banjoist, and a drummer who can initiate a soft-shoe dancer, then look no further. Next try his "Ragtime Piano Hits (GA209 SD)".

Collette & Counce: Jazz On The Bounce
Bel Canto SR1004

In addition to its stereo demonstration record, Bel Canto makes its entry into the field of stereo discs with a series of five introductory albums. They were prepared for tape release and the economics of their transfer would seem to work out to average the playing time of two tapes for about two dollars less than the price of one. The single item directed primarily to a jazz audience combines two West Coast sessions. Curtis Counce, a superior bassist, heads a quintet which shows its Parkerish inclinations and aptitudes on *Chasing the Bird, Move*, and *Head Gear*. Leisurely switching from flute to clarinet and saxes, Buddy Collette plays *Bass Rock, Soft Touch*, and *The Monster*. His quintet boasts an excellent guitarist, possibly Barney Kessel.

Collette contributes also to a mood album "Aloha To Jazz (SR1002)," paying a South Sea Island visit of nearly nine minutes on *Blue Sands*, a rewarding excursion for his drummer. Another original, the uptempo *It's You*, speeds them on their journey. The crash of surf surrounds a more conventional Hawaiian vocal group, The Polynesians, in a sheaf of hulas and the gently nostalgic *Beyond the Reef.*

"Beer Barrel Polka Time (SR1003)," is paced to resounding tuba as Razzberry Reynolds unfurls ten lively polkas, including the inevitable round indicated in the title. A glockenspiel enlivens *Billboard March* and if this band fails to arouse your enthusiasm for bouncing tempos, then nothing will. Anyway, try holding your beer in front of that tuba. It may blow off the foam.

"Listening To Larry (SR1005)" is best characterized by the fact that Larry Fotine spent two years arranging for Lawernce Welk. He leads an orchestra of thirteen pieces on a dozen tunes, dedicated to shuffle rhythm, a smooth sound and all eminently danceable. *Bye Bye Blues, At Sundown*, and *Sleepy Time Gal* are on the list.

"Resort Favorites (SR1006)," presents Harry Marshand conducting an orchestra in a typical society dance medley. Twenty-three numbers are outlined in short order and flow suavely along to unflagging tempos.

Under the supervision of Ted Kloba, the sound is uniformly good. The pressings on transparent plastic are excellent, but the liner notes are vague and uninformative. Recorded tapes are packaged frequently without vital statistics concerning personnel and content. It is hardly wise to continue the practice when up against greater competition.

Marty Grosz: Hooray For Bix!
Riverside RLP1109

A group of midwestern musicians, assembled briefly under the *Honoris Causa* Jazz Band banner in honor of Bix Biederbecke, plays a dozen tunes recorded by the cornetist at various stages of his eventful career. The guiding spirit is the guitarist Marty Grosz, whose father is the artist George Grosz. With Carl Halen, of Gin Bottle Seven fame, he wrote settings for small band of several numbers remembered only in the heavy arrangements used by Paul Whiteman and Jean Goldkette. The Bixian horn may be stilled, but there is no little gratification in listening to *Lonely Melody, Changes, Clementine*, and *Love Nest* for more than the cornet chorus.

Halen is joined on *My Pet* and *Because My Baby* by the second cornet of Turk Santos, who turns to guitar for a touching solo on *Oh, Miss Hannah*. Tut Soper plays a healthy piano, Harry Budd is on trombone, and two vocals by Grosz cause no pain. The stereo effect is enhanced by two clarinetists alternating on saxes. Bob Skiver, of Russell-Teschmaker lineage, switches to tenor, while Frank Chace hews a New Orleans line and doubles on baritone. Stereo separation keeps the rhythm from sounding bunched or chunky. The recording was made last December at Yellow Springs, Ohio, by Dave Jones, formerly engineer for his Empirical label, and indicates that the musicians of the region still have him as a mentor.

Ken Davern: In The Gloryland
Elektra 201-X

Headed by the youthful Ken Davern, a twenty-three-year-old disciple of the veteran New Orleans clarinetist George Lewis, this gathering comprises to all intents and purposes a reunion of the Red Onion Jazz Band. One of the few revivalist groups based in Manhattan, it was led by Bob Thompson who resumes his seat as drummer, a post he fills in addition to teaching psychology at Columbia University. Bassist Arnold Hyman teaches school in Philadelphia, and trombonist Steve Knight is another psychologist. Frank Laidlaw plays cornet and Carl Lunsford, a graduate of Gene Mayl's band and the musical "The Boy Friend," is on banjo.

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STEREOPHONIC

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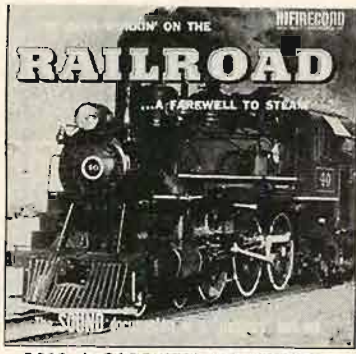

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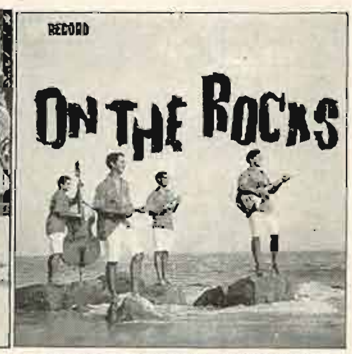

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and *Precious Lord*, they are to be commended for the sincerity of their approach. More frequent reunions should dispel a certain stiffness in the non-secular numbers. It is the one jazz item in Elektra's initial stereo release not available previously in monophonic form. The engineering by Dave Hancock brings the sound into sharp focus and the mastering is credited to Rudy Van Gelder.

Duke Ellington: Black, Brown and Beige
Columbia CS8015

Fifteen years have passed since this composition was hurriedly finished for a Carnegie Hall concert, and it is still not completely recorded. One consolation in the delay is that Mahalia Jackson consented at long last to sing with a jazz orchestra. Another is the timing of its release to coincide with the introduction of the stereo disc. For it was Ellington who pioneered many of the features, now routine with big bands, best heard in stereo. His men developed the call-and-answer of muted horns to a fine art, and the tonal strength of Harry Carney's baritone sax was added more than thirty years ago. Finally, it comes at a time when his organization is undergoing one of its periodic upsurges of creative activity which may be reflected in critical and popular acclaim a year or so from now.

The first section, *Work Song*, begins to a foreboding beat on Sam Woodyard's drums before a vigorous statement by full orchestra expresses the curiosity of a people set down in a strange land, explored by Harry Carney, Shorty Baker and Quentin Jackson in their solos. *Come Sunday* is introduced instrumentally, and then the two themes are set forth together by the stimulating horns of Cat Anderson and Britt Woodman against the entire voice of the band. Like the best of Duke's work, his writing for Miss Jackson is a new experience in music. She sings with religious intensity and deep insight, humming a tender final chorus.

In the main, her accompaniment is piano, bowed bass and Ray Nance's violin, and the sudden shift from full band presents the engineers with a knotty problem. Columbia seems to bring its "black box" into play to solve it, and in stereo her voice appears to move forward from the left at the height of a phrase to the center and recede on the right. Recorded in Hollywood, the monophonic version is most successful, but stereo holds more interest and plays back well on a single channel. A violin interlude restates the theme before Miss Jackson concludes by singing the *Twenty-third Psalm*.

MONOPHONIC

Big Bill Broonzy: The Blues
EmArcy MG36137
Jesse Fuller Good Time Jazz L12031

Big Bill put his life into his songs, the simple everyday joys and trials of existence, and now he is gone. His death coincided closely with the release of this collection, dating from 1951. It was recorded when his blues were losing favor in this country, though gaining respect in Europe, and disappeared into the vaults. Unlike his two volumes for Folkways which allow him the leeway of an LP, these dozen tracks average three minutes each and are representative of most sides made during his thirty-odd years of visiting studios. He was a master of the art of tailoring his stories and building a climax to fit this time limit. On eight country-styled numbers, his guitar is reinforced only by a bass player or pianist. In the company of a pair of primitive saxophones, drums and the strong blues pianist Bob Call, he sings *Southbound Train, Leavin' Day, You Changed*, and *Tomorrow*. Here his style is about as urbanized as it ever became.

Big Bill is quoted in the notes as declaring, "Maybe the blues'll die someday. But I'll have to die first." He would have received heartening reassurance of their vitality from the first recorded appearance of 62-year-old Jesse Fuller, folk artist extraordinary now residing in Oakland, California. A singer, composer and twelve-string guitar player, he is also somewhat of a one-man band, utilizing instruments of his own devising. He welded his set of foot cymbals, fitted his kazoo and harmonica to a harness, and invented the fotdella, an ingenious substitute for a stringed bass. In addition, he is likely to hum a trombone solo, on *I'm Going To Meet My Loving Mother*, or vocalize a drum break, on *Tiger Rag*, and his rhythmic pulse is unflagging on the difficult *Fingerbuster*. His treatment of traditional material is highly original, but the stamp of authenticity is on these and his own compositions. These include a prison song, *99 Years*, and *Memphis Boogie*.

Although Jesse and Big Bill never met, Leadbelly once paid the former a visit and he is well equipped to sing their blues. When producers of television jazz shows get around to investigating the origins of their subject, they can do no better than search out Fuller at his workaday task of operating a jackhammer in the Bay Area. Meanwhile, the blues will live by courtesy of a few perceptive record companies.

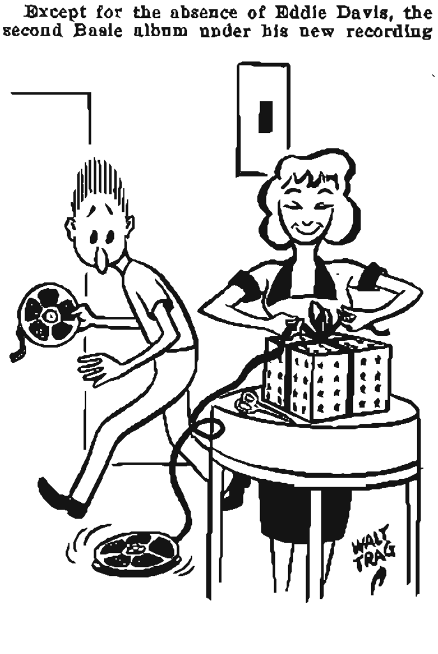
The Banjo Kings, Vol. 2
Good Time Jazz L12029
Red Nichols: In Love With Red
Capitol T999

The jazz of the 20's was not always raucous and tried to avoid the razz-ma-tazz featured on some present-day commercialized recreations of the era. Few are more aware of the fact than the musicians who gravitated to the relative security of the Hollywood studios. Since 1951, a Los Angeles group known as The Banjo Kings has demonstrated the startling possibilities of one of our few native instruments. Dick Roberts and fellow monarch, Red Roundtree, may yet save it from becoming extinct. Six selections are reclaimed from a 10-inch LP, and eight items recorded last year extend it to standard size. On these last, Vito Mumolo adds the voice of his tenor banjo and the pianist is Stan Wrightsman, with Country Washbourne or Ray Leatherwood on bass, and Nick Fatool on drums. Their expression of the sentiments of *School Days, The Band Played On*, and *I Want a Girl* are skilled and pleasantly tuneful.

Almost as much an American institution as the banjo, Red Nichols aims his latest effort at the popular market, perhaps in preparation for the motion picture of his life story, and augments his Five Pennies with a chorus and a dozen strings. He plays his signature, *Wail of the Winds*, and John Scott Trotter's theme for the George Gobel show. Joe Rushton's bass saxophone enlivens *Manhattan Rag*. The arrangements by Heinie Bean and Bobby Hammack, who provides a framework for the leader's trumpet on *Bugler's Lament*, keep the strings under control and the music is light and made for dancing.

Basie Plays Hefti Roulette R52011
K. C. In The 30's Capitol T1057

Except for the absence of Eddie Davis, the second Basie album under his new recording



alliance is almost a sequel to the first. As before, Neal Hefti furnishes tasty vignettes to highlight a soloist or particular set of instruments against the rich ensemble sound of the band. Frank Wess draws a major assignment and his flute dances lightly through *Cute, Scoot*, and *Late Date*, each a fetching exercise as concise as its title. The trombonists are led by Al Grey on two numbers, and the tenor saxes are featured in a chase episode by Billy Mitchell and Frank Foster on *Count Down*. Trumpeter Snookie Young tenderly explores the slow, balladic *Pensive Miss*, and crackling riff tunes spot Joe Newman on *Sloo Foot*, and Thad Jones on *Pony Tail*.

The Count restricts his piano to his usual canny phrases and prefatory bits. There is, in fact, considerably more Hefti than Basie in this collection. The band is currently in an orderly state of flux, gradually changing its style and widening its scope. It seems odd to find Hefti, who wrote his best arrangements while simulating the sparkling heads created by early Basieites, drawing it closer to Broadway and the more conventional swing settings. He is more likely to go back to the *Jeepers Creepers* figure of *A Little Tempo, Please*, than the blues. Where a resolute rhythmic pulse once drove the band, it is now anchored to sleek, carefully-played riff patterns. That they succeed so well is a tribute to Hefti's ingenuity, Sonny Payne's exciting drumming, and the brilliant section work. Any of the earthy qualities which once distinguished the band comes from individual soloists at this point. The recording is clear and embraces the group's dynamic force almost as fully as its predecessor.

For those persons desiring to recall the origins of the band in the Kansas City of the 1930's, Capitol restores a dozen sides from the period. Jay McShann plays *Moten Swing*, and Bus Moten presents *It's so Hard to Laugh or Smile*, the lovely theme he wrote with his brother. Julia Lee sings four numbers and Joshua Johnson delivers his haunting *Days*. Also heard are Tommy Douglass, Jessie Price, Crown Prince Waterford, and Charlotte Manfield. The only track suffering overly from the age of the recording is by Walter Brown, accompanied on tenor sax by Ben Webster. John Cameron Swayze, a reporter in the city at the time, delves into his memories on the liner and traces the careers of some of the artists. A valuable album for Hefti, or anyone else.

Andre Previn & His Pals: Gigi
Contemporary C3548

The fourth modern jazz reading of a musical show by Andre Previn and cohorts differs from its predecessors in that the pianist was involved with the Lerner and Loewe score from its inception. As musical director of the film, he was responsible for a number of arrangements, adapted background music, and conducted the studio orchestra. Such familiarity, in this case, breeds a certain insight and only the ballads remain close to the original. *Gigi* is revealed in the dimly-veiled hues of a Marie Laurencin painting, and *I Remember It Well* is colored in the impressionistic hues of the period.

Some of the best moments come from the expansion of transitory themes into sustained compositions. *A Toujours*, a background piece, becomes an exhilirating vehicle for bassist Red Mitchell. Previn preempts *Aunt Alicia's March* for himself, improvising thirteen challenging blues choruses to make it a highpoint of the session. His classical training enables him to play the sweetest dissonances this side of Stravinsky. Shelly Manne's brushwork moves *The Parisians* to a Basie rhythm, and *Thank Heaven for Little Girls* is swung lightly. Like the other shows in this series, the trio's treatment is needed to realize the possibilities of the score and the superb recording makes the experience a pleasure.

Chico Hamilton: Introducing Freddie Gambrell **World Pacific PJ1242**

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Fred Gambrell in a club two years ago, the drummer felt the urge to play behind him and returns to a trio format to assure an accomplished showcase for a fresh and imaginative talent. His rhythmic skill and subtle tonal coloration help put new finery on *Lullaby of the Leaves, These Foolish Things,* and *You're the Cream In My Coffee.*
Gambrell omits needless filigree and is best in his blues originals. He confesses to an admiration for other pianists from Albert Ammons to Oscar Peterson, but any overt influences seem thoroughly absorbed and channeled into a resourceful voice of his own. He believes in using both hands to full effect and the piano sound is recorded in excellent balance. Bassist Ben Tucker, somewhat of a find in his own right, contributes *Devil's Demise.*

Jonah Jones: Jumpin' With Jonah — Capitol T1039
The Fabulous Sidney Bechet — Blue Note BLP1207

This is the third release in a series which placed Jonah Jones' name on the bestseller lists in competition with sundry popular artists. There it is likely to remain, as a large portion of the general public recognizes in the muted tones of his trumpet an irresistible invitation to revisit a familiar tune. Shaped by an assortment of seven mutes, a multitude of shadings pour from his horn and make his style more vocal than that of most singers now in the limelight. His variations never stray far from the melodic line and are charged with a joyous emotion. The curious shuffle rhythm he likes to use on most numbers is lightened considerably this time by Hank Jones, his guest pianist. The qualities of a youthful Louis Armstrong characterize his singing on *Baby, Won't You Please Come Home, It's A Good Day,* and *A Kiss To Build A Dream On.*
Some members of his new audience may want to hear how he sounded, before grazing in such green pastures, in a sextet fronted by Sidney Bechet, the elder statesman of the soprano sax. In this restoration of a 1953 recording, Jones plays *Rose of The Rio Grande, Sweet Georgia Brown, All of Me,* and *Ding-Dong Daddy.* Sidney DeParis, another master of the mute, replaces him on the reverse side and Jimmy Archey is trombonist on both.

Jean Thielmans: Man Bites Harmonia! — Riverside RLP12-257
Dorothy Ashby: Hip Harp — Prestige 7140
Buddy Collette: Swinging Shepherds — EmArcy MG36133

Rather than mood jazz, an unfortunate term which might be implied by the odd instrumentations, these albums are more properly categorized as jazz to fit a mood. Even though he sublimates his baritone sax to underline the harmonica, the presence of Pepper Adams hardly suggests a setting for casual listening. Together, their sprightly interplay demands a period of undivided attention and you will save this record for just those moments.
Best known as guitarist in the George Shearing Quintet, Theilmans plays that instrument on *Imagination, 18th Century Ballroom,* and doubles on *Soul Station,* a functional blues original. On harmonica, he ranges through trumpet styles from *Strutting with Some Barbecue* to *Don't Blame Me.* The unusual blend fails to faze the rhythm section of Wilbur Ware, Kenny Drew, and Art Taylor, or the engineer, Jack Higgins.
Dorothy Ashby, a young harpist from Wayne University, has led her own trio during the past three years. In her record debut, she is aided immeasurably by Frank Wess, whose multiple skills grace the Basie band. His flute adds spice to *Dancing in the Dark, Charmaine,* and *There's a Small Hotel.* He might have advised her that interpolated quotes, however attractive in a club, wear thin after repeated hearings. But more experienced artists forget this, and the harpist supplies two rewarding blues, using a singular guitar-like effect. Herman Wright, her regular bassist, and Art Taylor complete the quartet.
Buddy Collette, apparently not satisfied with a single flutist as teammate, assembles a quartet of Bud Shank, Paul Horn and Harry Klee, plus rhythm, to play a variety of flutes and a piccolo or two. They work out on a number of originals, Pete Rugolo's Machito, and two improvisations done on the spur of the moment. It will take you considerably longer to sort out the soloists.

Harry James: The New James — Capitol T1037
Terry Gibbs: Plays The Duke — EmArcy MG36128

Two tributes to jazz personalities are signified on these albums, the one openly stated, and the other implicit in the light trend of the new James organization. It declares an allegiance to the principles of pulsing rhythm and warm ensemble attack laid down by Count Basie. The bravura James trumpet style is subdued into lean, muted solos that prepare thoughtfully for virile flights by Willie Smith on alto sax and Sam Firmature on tenor sax. The tried backgrounds of his former routine are replaced by a complete new book from the pens of Ernie Wilkins, Jim Hill, Bill Holman and Neal Hefti. Drummer Jackie Mills and guitarist Dennis Budimir give a firm thrust to the tempos, guiding them away from the stolidity which hardens the arteries of so many bands. This one swings all the way.
Terry Gibbs and Pete Jolly unite to express a few personal ideas on ten themes familiar to all Ellington fans. In addition to vibraharp and xylophone, Gibbs plays marimba for the first time on records. The deeper sound of its rich, wood tone blends nicely with Jolly's accordion and offers a pleasant contrast to his fleet vibes. With bassist Leroy Vinnegar feeding the soloists, they move with a compact assurance that comes from a thorough knowledge of their subject and each other. Gary Frommer, on drums, completes the quartet.

The Four Freshmen In Person — Capitol T1008
Sarah Vaughan: At The London House — Mercury MG20383

A college audience seems to put performers at ease and some of the most successful on-the-spot recordings come from campus auditoriums. The Four Freshmen are in their element on stage at Compton Junior College and enjoy a relaxed outing in the company of the assembled students of the California institution. A pair of microphones set up for stereo is the only confining force in evidence. As several of their previous hits are renewed during the concert, it is likely to appear in due course with an added dimension in sound. The fine acoustics of the hall contribute immeasurably to the trumpet solos of Ken Albers and the lofty trombone passages of Bob Flanigan. On *Sweet Lorraine,* the comic vocal by Ross Barbour seems more diminutive than ever, and Don Barbour dramatizes the theme of *Old Folks.* By the time the fourteen numbers are concluded, the quartet's popularity among those youngsters who have outgrown rock and roll is readily understandable.
An invited nightclub gathering can be one of the worst trials an artist is asked to face and Sarah Vaughan's skill in contending with this handicap, among others, adds spice to her appearance at Chicago's London House. Thad Jones, Wendell Culley, Henry Coker and Frank Wess from the Basie band join her regular trio on a set which begins with *Like Someone In Love,* followed by a premonitory *Detour Ahead.* Before she is through, the singer requires her wide experience and knowledge of style to calm her listeners, employing all her wiles on *Speak Low.* A good deal of her customary polish rubs off in the process, to be replaced in these parts by an increased respect for her talent as a performer, and the audience is not likely to forget her sentiments on the closing *Thanks For The Memory.* Listed on the liner is the microphone placement, used by Malcolm Chisholm of the Universal Recording Corporation, and a stereo version is forthcoming.

Arthur Lyman: Bwana ā — Hifirecord R808
Webley Edwards: Fire Goddess — Capitol T1033
Alfred Apaka: Dreams Of The South Seas — Urania UR9016

Since the quest for exotic sounds turned record companies toward Hawaii, the variety of musical groups on the island are being brought into closer focus. These three conflict only in their point of origin and use of native instruments. As many purchasers of his "Taboo" album know, Arthur Lyman loves to seize upon an unlikely theme to bring under the spell of his talented quartet. Aware of the devices of modern jazz and both Eastern and Western music, he mixes them discreetly on such disparate numbers as Schubert's Serenade, *La Paloma, Malaguena,* and *Colonel Bogey's March.* Joining the group this time are Japanese vocalist Ethel Azama, pianist Paul Conrad and Chew Hoon Chang, a septuagenarian artisan on butterfly or moon harp and bamboo flute. But there is no lack of native sounds on the title song, and David Kupele's *South Pacific Moonlight* employs conch shells and ocean surf. As before, the perfect acoustics of the Kaiser Aluminum Dome contribute to a truly exceptional recording.
Webley Edwards stays close to the islands in a compilation of ancient chants and the more recent *Beauty Hula, Thoughts of Love,* and *Chant of the Islands.* The sixteen piece Hawaii Calls Orchestra, supplemented by a host of singers and chanters, features sharkskin drums, gourds, lava rock castanets, conches, moon harp and gongs. Besides authentic tributes to Pele, the Fire Goddess, there are *Canoe Song, Tapa Dance,* and *The Old Church.* Made in the Princess Kaiulani Meeting House, on the beach of Waikiki, the recording benefits from the large hall and the miking is just right for the size of the group.
Alfred Apaka, who strayed from the islands to become an old hand at entertaining in hotels and supper clubs, sings a dozen numbers in the romantic style so popular in those surroundings. There is greater emphasis on steel guitars and ukuleles in his arrangements and a chorus fills out the backgrounds.

Johnny Puleo And His Harmonica Gang, Vol. 2 — Audio Fidelity AFLP1859

The tight little band of virtuosos gathered about Johnny Puleo devotes a second album to the cause of the lowly mouth organ. In elevating it to concert status, they are shepherded by an energetic leader who combines the high humor of a pixy and a complete mastery of his instrument. He views a tune with the practiced eye of a seasoned variety artist and packs his arrangements full of the surprises at the command of a veteran showman. The compositions of Strauss, Sousa, Moussourgsky, and Doucette are all grist for his mill and *Sabre Dance* is played to the hilt. Spanish rhythms are introduced on *Espana Cani,* followed by the colorful *Granada.* When not toying with the textures of a large orchestra, the ensemble enlivens several popular tunes. If you harbor no partiality for harmonicas, it may well be that you never heard them played like this.

Swiss Mountain Music — Capitol T10161

A conservation program for the massive alphorn, said to be rapidly becoming extinct in its native Alps, should excite the fancy of some exponential horn designer. Its construction enables it to float a clean, pure tone across mountain tops and over the bells of grazing cattle. It introduces a varied assortment of Swiss country dance music, gathered from several different Cantons and interspersed with the enthusiastic cries of strong-lunged yodellers. Their habit of adding a rhythmic pulse by swishing large silver coins about in an earthenware basin lends another unusual sound. The music is lively, full of traditional good humor and relatively unspoiled by the inroads of present-day communication channels. It is claimed that the alphorn was tracked to its lair with modern sound equipment for the first time and it performs nobly. Æ

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• **Atlas Cone-Projector Speaker.** Engineered especially for use in a wide variety of applications such as extension speakers for juke boxes, industrial music systems, and outdoor installations of all types, the Model W-6 is an ideal choice where good musical reproduction is desired but where

the ultimate in wide frequency range is not necessary. Power rating is 15 watts, frequency range is 140 to 8000 cps, and dispersion is approximately 120 degrees. Dimensions are 15"w × 12"d. The W-6 is finished in beige baked enamel, with a contrasting light brown plastic rim deadener to minimize the effects of horn resonance. Atlas Sound Corporation, 1451 39th St., Brooklyn 18, N.Y. **M-2**

• **Grommes Stereo Preamplifier.** Designed to operate with two high-quality power amplifiers, the Grommes 208 unifies and controls the separate components of a stereophonic hi-fi system. Circuitry con-

sists of two independent channels for each of the five inputs with ganged selector, turnover, roll-off, volume, bass and treble controls. A function switch is included to enable channel A or B to be used as a monophonic preamp driving both power amplifiers when no stereo program source is available. Variable equalization is afforded on both channels for stereo records. Feedback circuits are used throughout for optimum frequency response and low distortion. Self-powered with d.c. on all filaments, the 208 is ideally suited for operation direct from tape heads. Beautifully styled in charcoal gray and brass, it is adapted for either table-top or cabinet installation. For complete specifications, write Grommes Division of Precision Electronics, Inc., 9101 King St., Franklin Park, Ill. **M-3**

• **Ferrograph Stereo Recorder.** Introduced originally as a professional instrument, the new Ferrograph Stereo 88 is now available for home use. It affords both stereo record and playback, as well as half-track monophonic operation when desired. Dual operating speeds are 7.5 and 15 ips. Three motors include a hysteresis synchronous unit for capstan drive and two shaded-pole

types to handle rewind and fast forward. Frequency response is 40-15,000 cps ±2.0 db and wow and flutter are below 0.2 per cent. A switched VU meter is used for monitoring both channels. Matched recording and playback amplifiers are built to professional standards. For additional information contact the Ercona Corp. (Electronics Div.), 16 W. 46th St., New York 36, N.Y., sole U.S. agents for the British Ferrograph Co., Ltd., manufacturers of the Stereo 88. **M-4**

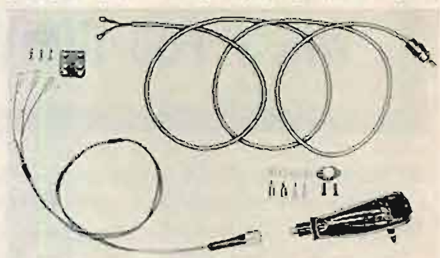
• **Glaser-Steers Stereo Changer.** Many unique features are incorporated in the new Model Seventy-Seven Stereo record changer. An ingenious quick-change cartridge holder permits instant changeover from a stereo to a monophonic cartridge with just the turn of a knob. A stereo-monophonic switch on the changer deck provides full stereo in one position, while in the other it directs sound from a mono-

phonic record to both speakers of the stereo system. A dual-channel muting switch silences output at all times except during the actual playing of a record. A new service receptacle below the changer deck permits automatic amplifier shut-off after the last record has been played. Rumble has been reduced with an improved 4-pole hum-shielded motor, with dynamically balanced rotor. Wow and flutter are negligible. Retained in the new stereo model are features which characterized the original Glaser-Steers monophonic model as an outstanding changer. For additional information, write to Glaser-Steers Corporation, 20 Main St., Belleville, N. J. **M-5**

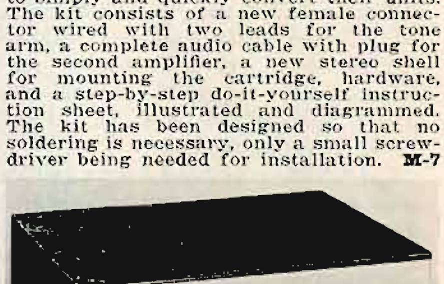
• **Tape Accessories Kit.** Both operation and maintenance of home tape recorders will be facilitated with the items included in this kit. Marketed by Robins Industries Corp., 36-27 Prince St., Flushing 54,

N. Y., the TK4-STD kit contains a "Gibson Girl" standard tape splicer, tape threader, head cleaner, tape cleaning and lubricating cloth, tape clips, reel labels, and 24-page booklet on tape editing and splicing. It may be purchased from dealers in tape and tape recorders. **M-6**

• **Garrard Conversion Kit.** All recent Garrard record players may now be converted within minutes to stereo wiring, using the new stereo conversion kit recently announced by Garrard Sales Corporation, Port Washington, N. Y. Although all new Garrard changers and players are now fully-wired for stereo, this kit will enable current users of Garrard models RC-88,



RC-98, RC-121, RC-121/II, and T-mk-II to simply and quickly convert their units. The kit consists of a new female connector wired with two leads for the tone arm, a complete audio cable with plug for the second amplifier, a new stereo shell for mounting the cartridge, hardware, and a step-by-step do-it-yourself instruction sheet, illustrated and diagrammed. The kit has been designed so that no soldering is necessary, only a small screwdriver being needed for installation. **M-7**



• **Madison Fielding Multiplex Converter.** Manufactured under license from Crosby Laboratories, the MX-100 multiplex converter will permit the home audience to receive stereophonic FM broadcasts from stations utilizing the Crosby system of

It is clear from the subscripts on R and C that Eq. (*18*) applies to the low-pass filter and Eq. (*19*) applies to the high-pass filter. Any arbitrary filter design which satisfies the appropriate expression Eq. (*18*) or Eq. (*19*) will afford 3 db voltage attenuation at the crossover frequency $f_c = \omega_c/2\pi$. If these general formulas are restricted by requiring $R_2 = aR_1$, $R_4 = aR_3$, $C_1 = aC_2$ and $C_3 = aC_4$. Eqs. (*18*) and (*19*) may be written in the form

$$y^2 + \left(2 + \frac{4}{a} + \frac{1}{a^2}\right) y - 1 = 0 \qquad (20)$$

Here

$$y^{1/2} = \omega_c R_1 C_1 \qquad (21)$$

for the low-pass filter, and

$$y^{1/2} = \frac{1}{\omega_c R_3 C_3} \qquad (22)$$

for the high-pass filter. The solution of Eq. (*20*) germane to this discussion is

$$y = \left(\sqrt{2 + \frac{4}{a} + \frac{5}{a^2} + \frac{2}{a^3} + \frac{1}{4a^4}}\right) - \left(1 + \frac{2}{a} + \frac{1}{2a^2}\right) \qquad (23)$$

If $a = 10$, Eqs. (*21*) and (*23*) yield

$$C_1 = \frac{1}{10.45\, f_c R_1} \qquad (24)$$

Similarly, if $a = 20$,

$$C_1 = \frac{1}{10.11\, f_c R_1} \qquad (25)$$

Clearly, the formula

$$C_1 = \frac{1}{9.7\, f_c R_1} \qquad (26)$$

appearing in the design data summary for the two-section low-pass filter, *Fig.* 2, is the limiting form of formulas like Eqs. (*24*) and (*25*). In other words, the larger the multiplier a the more accurate Eq. (*26*) becomes.

If $a = 10$, Eqs. (*22*) and (*23*) yield

$$C_3 = \frac{1}{3.77\, f_c R_3} \qquad (27)$$

If $a = 20$,

$$C_3 = \frac{1}{3.91\, f_c R_3} \qquad (28)$$

The limiting form of formulas like Eqs. (*27*) and (*28*) is

$$C_3 = \frac{1}{4.06\, f_c R_3} \qquad (29)$$

This expression is given in the design data summary for a two-section high-pass filter (*Fig.* 2).

2. Three-Section RC-Filter

In the three-section RC low- and high-pass filters shown in *Fig.* 1, let

$$\left.\begin{aligned} R_3 &= aR_2 = a^2 R_1 \\ C_1 &= aC_2 = a^2 C_3 \end{aligned}\right\} \qquad (30)$$

and

$$\left.\begin{aligned} R_6 = aR_5 = a^2R_4 \\ C_4 = aC_5 = a^2C_6 \end{aligned}\right\} \qquad (31)$$

Then if

$$y^{1/2} = \omega_c R_1 C_1 \qquad (32)$$

for the low-pass filter, and

$$y^{1/2} = \frac{1}{\omega_c R_4 C_4} \qquad (33)$$

for the high-pass filter, it can be demonstrated that y satisfies the following equation:

$$y^3 + \left(3 + \frac{8}{a} + \frac{2}{a^2} + \right) y^2 + \left(3 + \frac{8}{a} + \frac{10}{a^2} + \frac{4}{a^3} \quad \frac{1}{a^4}\right) y - 1 = 0 \qquad (34)$$

For each value of the multiplier a (which is a pure numeric) there is one positive root y which always satisfies $0 < y < 0.26$. Equation (34) may be solved graphically or by the use of the method of Horner, or Newton. Possibly the simplest procedure is to guess the value of y for a specified value of a, and verify the accuracy of the guess by substituting the values back in Eq. (34). A desk calculating machine is useful in carrying out this work. To insure an "intelligent guess," the writer has prepared a plot of y as a function of a. This graph is shown in *Fig.* 4.

The following procedure is suggested for completely determining the exact circuit values for a three-section RC crossover network:

(1). Choose the value of a. A reasonable range is $1 \leq a \leq 10$.

(2). Accurately determine y for the selected value of a, using Eq. (34), beginning with the y *vs.* a plot given in *Fig.* 4.

(3). Arbitrarily select the values of R_1 and R_4, keeping in mind "rule" (c), set forth at the beginning of this paper.

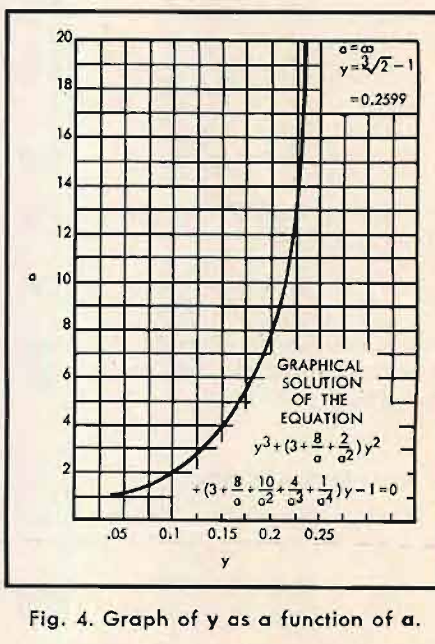


Fig. 4. Graph of y as a function of a.

(4). Choose the cross-over frequency $f_c = \omega_c / 2\pi$.

(5). Determine the value of C_1 from Eq. (*32*) and C_2 from Eq. (*33*).

(6). Complete the filter design, using Eqs. (*30*) and (*31*).

Discussion and Conclusions

In Part I an approximate solution of the RC high-impedance crossover network problem was presented. If the multiplier a is equal to or greater than 10, the formulas for the circuit values based on the simple minded approach to RC filter theory will be accurate enough for most purposes. Part II was written for the perfectionist. The only assumption made in the derivation of the formulas for the circuit parameters is that an infinite resistance terminates each filter, i.e., that the input circuits of the bass and treble amplifiers do not load their respective filters. This condition is approximated in practice if a cathode follower is interposed between a filter and the associated amplifier. (Refer to *Figure* 2 in reference 2.)

It is not necessary for the parameter a to exceed unity when designing a filter in accordance with the formulation of Part II, provided the load presented to the cathode follower driving the filters is not excessive.

Clearly, there are an infinite number of two- or three-section crossover networks that may be built for a given crossover frequency. There is no requirement that resistors of the same value be employed in the low and high-pass sections in "comparable" positions. Resistors of ±5 percent tolerance are readily available. Capacitors, except the more expensive ones, are ±20 percent tolerance. Filter components must be selected with care, and even so it will usually be necessary to pad each filter element to obtain the correct attenuation at the crossover frequency selected.

The question may arise as to whether to employ a two- or three-section dividing network in a given dual-channel amplifier system. The attenuation of a three-section high-pass filter at say an octave below the cross-over frequency is greater than that of a two-section filter at the same frequency. The low-frequency power handling capability of the treble driver may require one to select a three-section filter. A two-section high-pass filter should afford sufficient protection for the tweeter if a treble amplifier of sufficiently low power output potential is employed.[3]

Acknowledgment

The assistance rendered by Col. Thomas J. White, USAF, and Dr. Arthur Grad in the preparation of this paper is acknowledged.

[3] Charles W. Harrison, Jr., "High-quality treble amplifier." AUDIO, August, 1957.

LETTERS

(from page 6)

is based on measurements on twelve different types of FM tuners feeding a subcarrier receiver, and three different makes of background music receivers. By the compatible sum-and-difference technique, this 12-db difference is reduced to 6 db for the stereo channel. The result is a stereo performance very nearly equal to that of the main channel. In addition, full 15-kc fidelity is obtained on the subcarrier channel and the full stereo effect is obtained on the complete range from 30 to 15,000 cps.

Now let us analyze the effect of adding another subcarrier to this optimized system.[2] There are four effects which must be considered. They are:

1. Power loss due to division of the modulation percentage between the two subcarriers and the main channel.
2. Power loss due to limitation of the subcarrier frequency spectrum to thereby limit the frequency deviation on the stereo subcarrier.
3. Increase in the possibility of cross modulation between channels from the extremely simple single subcarrier situation where cross modulation is not a factor, to a situation where there may be six possible cases of cross modulation.
4. Curtailing the fidelity on the subcarrier channel so that the full stereo effect is not obtained over the 15-kc audio band.

The power loss due to the division of the modulation percentage between the two subcarriers of the main channel, is brought about by the requirement that the percentage of modulation applied to the main channel and subcarrier channels must be divided so that the total is not more than 100 per cent. The optimized standards that I have proposed apply a 50 per cent modulation to the subcarrier and 50 per cent to the main channel so that the total becomes 100 per cent. For the case of the two subcarrier channel system, due to the presence of the "foreign" program, it would probably be dangerous to put more than 30 per cent modulation on the sum total of the two subcarriers. This is the value which is now authorized by the Subsidiary Communications Authorizations. The danger would be brought about by the possibility of the foreign channel cross-modulating into either the stereo subcarrier channel or the main channel and thereby damaging either the monophonic or stereo transmission.

With a total possible modulation of 30 per cent applied to the two subcarriers, each one could only be modulated 15 per cent. Thus by this division of power among channels, the percentage of modulation on the stereo subcarrier channel has been reduced from 50 to 15 per cent, which is a loss in signal-to-noise ratio and power of 10.5 db with respect to the optimized system.

The power loss due to the limitation of the subcarrier frequency spectrum shows up as a lowering of the frequency deviation on the stereo subcarrier. There have been many claims and proposals that full 15-kc fidelity could be obtained on the second subcarrier. As I see it, these claims have overlooked the filter requirements which would ensue if an attempt were made to crowd in 15-kc audio fidelity. The filter to reject the background music should reject it to the extent of at least 60 db in the listener's receiver. Such a filter with such sharp cutoff requirements becomes complicated and expensive. It would be more practical to allow for an audio fidelity of about 7.5 kc on the second subcarrier and a frequency deviation of 7 kc. The loss of power by reducing the deviation on the stereo subcarrier from 25 kc to 7 kc is 11 db. This loss is to be added to the 10.5 db lost by the reduction of the percentage of modulation. The lowering of the audio fidelity from 15 kc to 7.5 kc lowers the noise 3 db so that the two-channel subcarrier system has a total loss compared to the single-channel optimized system of $11 + 10.5 - 3$ db, or 18.5 db. This is a power loss of 71 to 1 in power. In other words, if the FM transmitting stations wanted to cover the same area with a stereo transmission using the two subcarrier system, they would have to raise their power 71 times that of the single channel optimized system. If the power was not raised, the service range would be reduced to 69 per cent of that of the single channel system, and the service area reduced to 48 per cent. If the two-subcarrier-channel broadcaster were foolish enough to not take advantage of the power gain, compatibility, and balancing features of the compatible mixing system, the power loss would be increased from 71 to 1 in power to 284 to 1 in power. The service range would be reduced to 63 per cent and the service area to 40 per cent. These figures are all based on measurements on actual receivers and computations of the service range and area using the FCC charts in their Standards of Good Engineering Practice.

The increase in the possibility of cross modulation, brought about by the two-subcarrier system is an extremely dangerous situation because FM multiplex cross modulation is dependent on the characteristics of the receiver as well as the transmitter. The transmitter may be made perfect so that there is no cross modulation, but the cross modulation may be introduced by a maladjusted FM tuner. With a foreign program to contend with, as distinguished from the same program in the single-subcarrier channel system, the cross modulation jumps from a negligible situation to one which has the possibility of damaging the monophonic transmission and a larger possibility of damaging the stereo transmission. In the case of background music operation, special receivers can be employed which are adjusted by trained technicians. In the stereo situation no such control is possible. The experience of the background music operators, who have been plagued by cross modulation problems, would indicate that before such a system is set up, there should be a surety that the foreign program does not damage the monaural or stereo reception.

Curtailing the fidelity on the subcarrier channel is somewhat ameliorated by the compatible-sum-and-difference technique. Under these circumstances, assuming an audio fidelity of 7.5 kc on the stereo-subcarrier channel, since each loudspeaker receives half its energy from the main channel and half from the subcarrier in the compatible system, full 15-kc response is obtained on both loudspeakers. The upper half of the response is somewhat lowered to the extent of 6 db, which may be corrected by tone control, and is a non-stereo transmission. However that type of transmission is, of course, superior to no transmission at all in the region above 7.5 kc.

It has been argued that transmission of frequencies up in the higher frequency range above 6 kc is not necessary for stereo. I believe that this conception has been drawn from the old data where the stereo effect was obtained by measurements on the phase effect alone. I recall such an article in one of the early issues of AUDIO In a Bell Telephone Laboratories book, *Speech and Hearing in Communication* by Harvey Fletcher, it is pointed out on page 225 that the *amplitude* effect of the stereo response is strongest in the frequency range of 5 to 15 kc. Thus when the amplitude effect is considered, the complete range of frequencies becomes important. To bottleneck the future of any system which curtails the range of frequencies, is again to stifle progress. On the other hand, where conditions are such that it would be impossible for the listener to receive any form of stereo other than one having a nonstereo range above 7.5 kc, that stereo is superior to no stereo at all.

In concluding, I would like to point out that FM broadcasting is now in a period of renaissance. In the past two or three years, the high-fidelity music lovers have shown such an interest in FM broadcasting that the number of FM stations on the air is increasing at a rapid rate where at one time it was declining. Allocations are now saturated in several of the large areas, and where channels are available several applicants are applying. Programming practices are changing in that, instead of merely connecting the FM station to the AM channel, many stations are now separately programming their AM and FM stations. Sponsors are realizing the value of the FM broadcast, and are turning towards FM. Now, with the terrific acceptance of stereo by the public, as evidenced by the violent acceptance of the stereo disc, we have the possibility that, with FM being able to broadcast stereo, it will come into its own as a superior broadcasting medium. It is my feeling that, if both the optimized wideband single channel system, and the two-channel subcarrier system are allowed, the broadcasters on the two-channel system may come to a time where they will voluntarily wish to improve their stereo transmission and increase their coverage, and will discard the background music business as being no longer required in view of the increased possibilities of true broadcasting brought about by the desire of the people for good music and stereo.

MURRAY G. CROSBY,
Crosby Laboratories, Inc.,
Syosset, N. Y.

(We appreciate both of these letters and are pleased to be able to print them in full. The entire problem has many ramifications, ranging from the presumed public ownership of the air to the equity of arbitrarily endangering established businesses. We don't know all the answers, but it is probable that the FCC will find out most of them and evaluate engineering data together with the business aspects of the proposal before making a final decision. We often wonder how public "interest, convenience, and necessity" is served by many of the AM stations with their present programming and the extremely low ratio of program to commercials. ED.

[2] *We did not propose that stations* must use both *stereo and background subcarriers —only that that they could if they wished, even though it might degrade quality. We believe stations would choose which service they would provide on the basis of the kind of business they wanted to pursue.* ED.

AUDIO ETC.

(*from page 12*)

hall effect seems to come from the same narrow central bulge in space.

If your two speakers, say, are spaced ten feet apart, the sound of a symphony seems to come from an area perhaps four feet wide, huddled near the center point between the speakers. They, the speakers themselves, seem to have nothing to do with this, as mentioned above—they are well "outside" the sound-source area and appear to the eye to be dummies, not even operating!

Now this sort of center-blob image, mono sound out of two speakers, is not good for musical listening in most cases. It compresses the sonic picture, squeezes it into a too-narrow apparent source area—if not a point source, then a blob source.

A stereo recording from the same two speakers, you see, adds very clear side-elements to the center—sometimes too clear, as we know. With definite rightish and leftish stereo sounds, in or behind each of the speakers, the total sound-spread is very much widened and deepened, for the typical stereo effect. But for mono sound there is no side-sense whatever. Therefore—the sound is only a blob in the center.

So I find that to use just one speaker, directed or off to the side at an angle, for maximum width of reflection, gives a more natural and more musical feeling to monophonic reproduction than the mono sound from two speakers, in stereo formation. I don't always bother to move furniture for it but if I have some heavy mono listening to do, these days, I do tend to shift to one speaker, reflected.

How about reflecting the sound of *both* speakers, for the mono situation? Oddly, this doesn't help as much as you might think, because the central-blob mono image persists even when the two speaker beams are each dispersed by reflection. Wherever they mix, the ear instantly centers the apparent source between them. Thus the image in this dual reflected reproduction is still too much compressed into that center position, two speakers or no. With one speaker, the effect is quite different—there is no centering, if the reflections are varied enough.

Still, it's worth a try, this reflecting of both stereo speakers for mono sound situations. Given enough irregularity of reflection you may achieve a respectably wide apparent sound source, though it isn't likely to be as wide as the speaker spacing itself, even at best.

(I can hear somebody shouting—why not throw the two speakers deliberately out of phase; then the sound will jump out to the end positions for plenty of width. True, but out-of-phase dual speaker reproduction is false in too many other ways for this to work.)

So there you have it, as of now. My earnest suggestion to all you hi-fi experimenters, in-business and hobby-style, is to keep your best mono equipment on hand and usable, alongside your stereo. Slide-in cartridges or plug-in heads (with wiring arranged to fit both systems), plus easily moveable speakers, will do the trick. Abandon your mono equipment only when you are convinced you can get along without it, by test. I'm not—yet. Next year, maybe.

ESL

I've had the loan of a number of the new stereo cartridges lately and wish I were engineer enough to say more than I can about their specific technical features. As usual, it is listening experience that counts in my case and that takes time.

The ESL was the second magnetic I received, after the Fairchild XP-4 (the one that at first seemed out of order—later turned out to be in excellent condition; it was external trouble). ESL makes a fairly expensive cartridge with a tricky watch-like precision action which I have not yet had time to study out; I'm too busy using the cartridge. My idea was to see what a really fancy professional unit, no expense spared, might be like for a sort of home laboratory standard set-up.

ESL went back to the use of transformers for this model, after moving out of that area in its latest mono cartridge (higher output). Probably required for best performance—it is that sort of thing that is necessary in a top-level quality development. Two transformers, German-made, and they work fine, even if they do cost in themselves as much as a complete mono-style GE reluctance cartridge, diamond-sapphire model. I don't mean to emphasize the cost, but if you try ESL you'll have to face up to it one way or another. Likely to be worth it.

With transformers, the ESL stereo cartridge gives a big wallop into your stereo preamplifier, enough to beat down all hum troubles due to low-level input. The transformers themselves can, of course, pick up hum and will do so if they come near to a good solid unshielded power transformer. But in most cases you won't run into trouble here. (You can always try moving them around in space near your amplifier, just to see.) I did have trouble in just such a situation, but all is OK now.

I also ran into noise pickup due to some species of then-unexplained group loop, but that, too, is now of the past. (These things usually do work themselves out with a little help.) Now that the ESL is down to business playing music, I find it a pleasantly smooth and reliable piece of apparatus, so far, after a couple of months' intermittent use. No troubles have developed, no harsh tone qualities nor bad tracking; if there's a jangle in the very high highs it is certainly far from obvious to my ears.

The unit is barrel shaped, an odd configuration, and it is rather heavy. (So are some other stereo pickups.) You may need extra rear-arm weight to balance it properly. But it takes moderate differences in stylus force rather easily, without noticeable change in response. (The vertical response is always highly sensitive to stylus force, for fairly obvious reasons.) Flexible in use, apparently rugged (so far) and nice in sound, what more do you want? It fills the bill, as far as I can tell, as an all around "laboratory" stereo pickup, and it should do as well in any hi-fi home.

The ESL transformers, I should note, are inconspicuous and relatively convenient; they are ready-cabled for instant insertion, with RCA plug and socket, and the transformer unit itself is hardly more than a big bump in the cable. (Yes—they are in phase. I tried that but quick!) You just hang them on the rear of your amplifier. Let 'em dangle well away from that power transformer, though.

Micamp

For use with the ESL and other similarly low-output, low-impedance devices including microphones, a neat little alternative to the transformer is the new Micamp transistor unit from Madison Fielding, a miniature battery operated affair with a gain of 30 db. It is only a little larger and heavier than the ESL transformer, eliminates any possible hum pickup, has a stated hum level of zero, draws .001 watt from its tiny cell and is supposed to run for a year or so without a new battery. The new battery goes in in a moment, with one screw, via a small panel in the bottom plate. No controls, just an on-off slide switch and two RCA-type phono sockets, in and out.

This little unit is also used as a mike "converter," to feed a low-impedance mike, at the end of any length of cable you can muster, directly into the usual high impedance mike input on tape recorders and so on. No wires, self-powered, and therefore mighty convenient as long as the battery holds out.

You may remember my write-up of Fisher's early entry into the all-transistor preamp field a good many seasons back. That was a bulkier affair, size of an old-style GE preamp, and it had a variety of inputs and outputs. It served me as an emergency mike preamp on my radio broadcast and continued in that use, on the original battery, for an entire winter, though I several times left the switch on for days by mistake. My only complaint was an occasional transistor "rustle" or irregular hiss; part of that noise was perhaps in a not-so-hot transistor that I didn't bother to look into. I would just bang the unit a couple of times and it would stop.

The Micamp is a streamlining of that early Fisher pioneer project, boiled down to minimum essentials and minimum size. So far—not a trace of background noise. It works. The sound seems fine and could well be according to specs, which claim it pretty clean. Won't do anything for such as Pickering, a cartridge with high-impedance output—but that isn't the intention. The thing provides gain for low-impedance units and that's that. Useful enough, though the cost isn't low, about $11 each, more than the ESL transformers. (But more versatile too).

One small outward criticism. Where ESL puts two short cables on its transformer, with male and female phono plugs, the Micamp comes without cables and its input and output are both female sockets. That's clumsy. It means making up special cables—two sets, for stereo—and it involves an undesirable reversal of direction. (A female output.) Yes, I know that customarily the recording output on most amplifiers is similarly female. There, a male cable would be hopelessly unwieldy. But on the Micamp, I suggest, a short output cable with male plug would save other users as much time as it would have saved me. Just plug directly into your preamp. Note also that for most professional low-impedance microphones a conversion will be necessary in the input plug. They normally use the larger "hermaphrodite" Cannon plugs. These are minor carpings and you won't take them too seriously. The Micamp is a neat gadget and useful.

Vico 87 and 89

Many readers will remember the somewhat caustic, if enthusiastic report in this column concerning the first all-transistor hi-fi amplifier, from Vico. It seemed to me that the Vico unit, full of almost naive "bugs" and miscalculations—the company had not been in hi-fi before though transistors were their speciality—was an important and interesting pioneer effort. I "bore down" fairly strongly upon the faults of that unit because it seemed, under the circumstances, a constructive move.

You'll be interested to know that the Vico people not only were pleasant about that fairly severe criticism, but thanked us for it and seemed only too happy to have an outside opinion that might add to their new know-how. It does happen! Criticism, even in the manufacturing field, can have positive results of usefulness, if the critic is lucky.

And so a couple of months ago, zoom through the mails came, not one, but two new Vico amplifiers, embodying improvements that are, I suggest, almost sensational for a Model Two in a totally new line from an inexperienced maker. The Vico 87 and 89 are semi-identical amplifiers, in the same decorative case and with the same controls; the 87 is a 20-watt model, the 89 a 35-watt job. They arrived just at the climax of my stereo hum troubles—in seconds I had violated my own just-announced ruling and had put them to work on stereo! After all—they were supposed to be humless as only an all-transistor amplifier can be. Everybody knows that. Moreover, with identical controls and general features, a tricky decorator case that stacks, so that one of them fits neatly on top of the other in interlock style (with air space provided for ventilation in between the two)—they really made a superb stereo set-up, even if I did have to use two volume controls. (They're similarly placed, and the settings are almost the same for a given volume, in spite of 20 and 35 watts —the sensitivity is about the same, quite aside from the rated power.)

So—I have only one mild complaint about these two all-transistor, tubeless amplifiers, which I have been using for stereo ever since then. They both produce hum. Not bad hum, but enough to annoy my sensitive ears, which are very easily annoyed on this score. Hum? Yes, even transistor appliances have it, alas. In Vico's case the main trouble seems to be in the phono equalization circuits; it is there that the hum is most noticeable. On the other inputs there is virtually none, even wide-open. (The mike input does some rustling; not too bad.)

But hum aside (and it hasn't stopped me yet), these are remarkable units. At the idle, with no signal input, they draw roughly 7 and 9 watts from the power lines. Efficient as all get-out. I tend to leave 'em on indefinitely. They are housed in a wood sided cabinet with a handsome white tilted front panel, framed in gold and marked in good looking and legible black design with red brand-name letters; the control knobs are clear plastic, with gold centers. High style, indeed, and I needed only to put a piece of red sticky tape on the two volume controls to put them in practical working order. (As I've muttered before, I do *not* like to have to fish for the volume control among a lot of identical knobs.) Quite decorative, at that —the red sticky tape.

These Vico models incorporate practically everything that was missing or lacking in the first model. Loudness contour control, the fixed type with several positions and OFF, a rumble filter (real low-down—I can't even hear the difference on most material), a three-position cut-off. Inputs in back cover all standard conditions, and—lo and behold—there are two radio inputs, one with a level-set; that, surely, was thanks to my suggestion, that on the first model the tuner input would be overloaded by the average tuner. (But on the input *without* a level-set this amplifier does not overload.)

There is a mike input, the usual transistor facility, taking low-impedance mikes without transformer. The over-all sensitivity is just right; the high-level input takes my Ampex signal beautifully, where the first Vico was dismally overloaded and not a thing to be done about it.

In other words—the Vico people have really gone into the hi-fi subject this time and have updated their working information to a remarkable degree in such a short time. These models, of course, retain the special transistor advantages, the low-impedance mike input and in particular the battery operation (12 volts) as an alternative to power line use. (You must still be careful of plus and minus.)

On battery, these surely mark a new level of top-quality luxury and power for portable, auto, camping, boating and general picnic hi-fi. In many a special installation, they should prove amazingly useful. Use them in your trailer, yacht, summer camp, auto, commuter plane, and your battery will hardly know the difference. But don't try to carry the 87 and 89 in your rucksack. These two amplifiers are now full-weight and full-size. They offer no great advantage in bulk and heft over most trim-type standard amplifiers of equivalent power.

Maybe Vico's next line will take advantage of transistors in the size aspect—a minimum complete hi-fi amplifier, say 8 or 10 watts, with basic simple controls and connections and light and portable like a camera. That ought to sell.

P.S. About that hum—I can't explain it and haven't asked the experts. Too bad— the amplifiers would be near-perfect, if it weren't for it. Not quite; one of them has its cut-off connections mixed up, the 12,000- and 8000-cps positions interchanged. Early-model slip, I expect. (The hum, too, probably.) I note too, while I'm at it, that the more powerful 89 seems to have a slightly higher noise level in the wide-open positions than the 87. Maybe this is merely the extra power. It also seems ready to oscillate or something at higher volume positions. Generally it impresses me as less stable than the 87, the 20-watter—that one is best.

Vico has a stereo model, equal to a pair of the 20-watt 87's in one case. Also a forthcoming FM-AM tuner that I'd like to see. All of these all-transistor, of course. The cost of these is in the middle-bracket range, centered near the $100 mark, the values well able to match the tubed competition in the same range, I would guess.

The Salad

I can't help occasionally appending a few of the novel uses I'm currently finding for hi-fi stuff. You may be amused and perhaps aided.

Take polyethylene or vinyl record bags. These make the most perfect vegetable or salad bags I know. Just dump food in them, pull the tops together and stick with freezer tape or a tight rubber band. Better than commercial freezer bags— they're bigger. I use them for deep-freezing assorted bulky foods, too. (My records don't suffer, I have several packages of extra bags—you can buy them in record stores.)

If you have any old Westminster Lab Series discs kicking around that are partially expendable, you own an even better gadget. Just dump the record out of that marvelous zipper plastic case and put it in an ordinary record bag. It'll be OK. Then—stuff the zipper thing full of portable salads and the like: they'll keep for days without wilting. The zipper is virtually waterproof, too, and my travelling salad stays fresh and doesn't spill. I've been carrying seads of sweet lettuce from my late garden down to the big city each week, for friends and relatives. Thanks, Westminster!

Suggestion: Fill a "polly" record bag two thirds full of water, bunch up and seal the top, sit the bag in your freezer compartment. In awhile you'll have a dandy hunk of rounded ice for a punch bowl. Just pour on hot water to loosen the plastic.

That's enough for this month. Æ

NEW SPEAKER SYSTEM SCORES IN CHICAGO, N.Y.

Two gentlemen from California appeared at the recent Chicago and New York hi-fi shows demonstrating a new speaker system. Because of its superior musical reproduction it received nods and smiles of approval and a repeated comment: "Best sound in the show".

This was the Audiosonic speaker system's exciting debut. It is the best because the two new components which comprise it combine to produce nearly perfect transient response.

The Audiolab Custom 16 Co-Axial Speaker: 2-way full range speaker with dynamic braking and a revolutionary polystyrene foam plastic cone. Wide, flexible suspension, 20-cycle resonance, cast frame, heavy magnet. Plus, the Anechoic Enclosure: beautifully styled new acoustical design. The first new commercially available enclosure principle since 1934. Sum total: The most truthful, stimulating musical sound outside the concert hall.

Thousands heard it at the shows and were thrilled. If you haven't heard it visit your local hi-fi dealer. Should he not yet have it in stock tell him to order so that the Audiosonic will be available in your community.

These fine products are also separately available. The Anechoic Enclosure comes in a variety of models and finishes. For Custom 16 technical information and comprehensive specifications, write to: **AUDIOSPEAKER LABORATORIES, P.O. Box 1082, Pomona, California.**

Circle 74A

PHASING

(from page 17)

tors. The axes of the two motors in a stereo cutter, such as the Westrex, are 90 deg. apart, both inclined 45 deg. to the record surface. It is common to indicate their motions as two vectors as is shown at (A) in *Fig.* 3. As shown, their phasing indicates a resultant motion in the vertical plane and none in the horizontal. If, instead, the vectors were shown as at (B), the resultant motion would be horizontal with none in the vertical plane. This is the correct phasing of the cutter to produce lateral motion from in-phase signals to the two recording amplifiers.

Another factor which leads to some confusion is that the right motor of the cutter causes modulation of the left groove wall and vice-versa. The R.I.A.A. Standard states that right side information appears as modulation of the outer groove wall. In the case of most pickups, motion of the outer groove wall generates an electrical signal on the left element of the pick-up. Here right and left are defined by facing the front of the cutter or pick-up in cutting or playing position. More specifically, motion of the outer side wall produces an electrical signal on the side of the pick-up nearer the center hole. Confusion on this point also arises from the Westrex 1-A Frequency Test Record which calls the outside groove modulation the left channel. The record is all right in itself, but is not labeled in accord with the definition of the R.I.A.A. Standard. This is

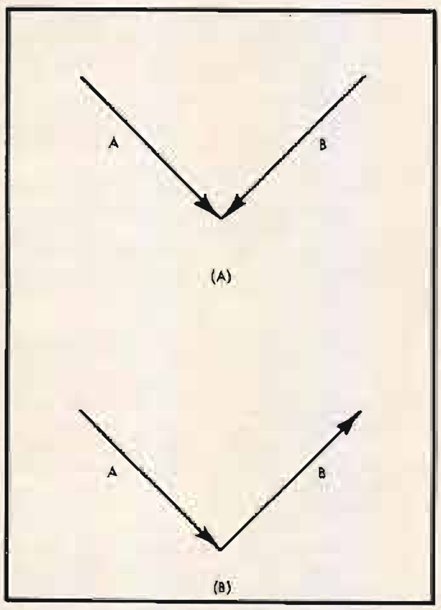


Fig. 3. (A) Vectors indicating the motions of the two channels in 45/45 stereo disc recording. With the phasing shown, equal motions result in vertical modulation only. (B) Here the vectors are phased so that equal motions on both sides will result in lateral modulation only.

understandable, since it was produced before the R.I.A.A. Standard was adopted.

The flexibility offered by stereophonic reproducing cartridges makes it possible to determine whether records have in-phase information correctly appearing as lateral information or just the opposite. By switching the output terminals, a good stereo cartridge may be connected to yield only lateral information or only vertical information. By reproducing a record alternately with these connections, the relative outputs may be compared. The correct connections of the pick-up cartridge may be checked by observing the results with a conventional lateral record. Of course, on stereo records which are sharply separated very nearly equal information will appear under the lateral and vertical reproducing conditions. All records having substantial third channel, or phantom, information as well as those made with the X-Y or M-S types of pick-up, should show greater energy laterally than vertically.

One favorable factor is that the improvement in reproducing equipment which stereo necessitates makes the results sound quite good even when improperly phased. This, however, does not justify any lack of effort to effect the best possible results. Æ

AIR SUSPENSION

(from page 21)

Relative radiated sound pressure expressed in decibels is:

$$SPL = 20 \log_{10} \frac{g^4}{\sqrt{\left[(g^2 - 1)^2 + g^2 \frac{S_T}{S_S} \right]^2 + \frac{g^2}{Q^2} (1 - g^2)^2}} \quad (9)$$

The equivalent circuit shows when the enclosure is tuned to speaker resonance, the impedance is a maximum and is resistive at this frequency so that the speaker excursion should be greatly reduced. In *Fig.* 5, (A) and (B) show the excursion of the speaker in a closed box and in a vented enclosure, respectively, verifying the intuitive reasoning. *Figure* 6 shows the effects of reduced excursion on the total harmonic distortion. Although speaker excursion is greatly reduced, the acoustic power radiated is not reduced. A plot of the ratio of port velocity to speaker velocity shows that for normal port Q's, (greater than 5), the acoustic power radiated from the port predominates over that radiated from the speaker cone for about ½ octave above and below speaker resonance. Since the port does not have a non-linear diaphragm suspension to create distortion, the over-all harmonic distortion is greatly reduced.

The port velocity for Q's of less than 5, as would be the case in distributed or friction-loaded ports, is of the same order as the speaker velocity and hence radiated power is greatly reduced. The cone amplitude is also increased and distortion is increased likewise.

The phase shift is very rapid above and below speaker resonance so that for all practical purposes the radiation from port and cone can be considered in phase above speaker resonance and 180 deg. out of phase below speaker resonance.

An examination of the expression for relative sound-pressure level shows that two quantities affect the output—speaker Q, and the ratio of enclosure stiffness to speaker stiffness. Decreasing the stiffness ratio and increasing speaker Q will increase output. It will be shown, however, that whereas the closed box required a speaker Q of about one for flattest response down to resonance, the same speaker in a vented enclosure of the same size now requires a Q of less than 1/2 for the same output. This allows using a higher speaker efficiency and/or higher damping factor in the amplifier, with an improvement in transient response.

Figure 8 shows the theoretical sound pressure response of a speaker in a vented enclosure for a stiffness ratio of 7, for several values of speaker Q. *Figure* 9 shows this experimentally.

This corresponds to a 12-in. speaker in an enclosure of about 2 1/4 cu. ft. volume. The flattest response corresponds to a Q of about 0.3. The response quite obviously is not flat down to speaker resonance. Consider, however, a specific example and compare this to the closed box response. Assuming a speaker resonance of 20 cps, *Fig.* 8 shows a flat response down to 54 cps for a Q of 0.32 with the 10-db-down point at 28 cps. The stiffness ratio of 7 would in the closed box case give a speaker resonance of 60 cps. *Figure* 4 shows that a speaker

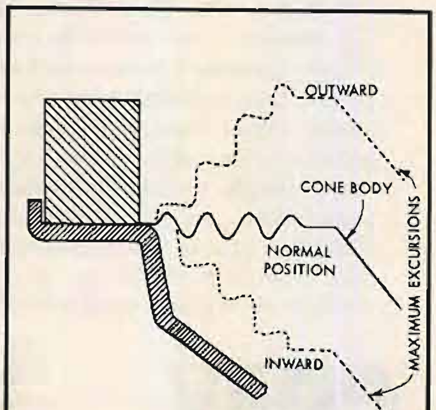


Fig. 11. Shape of Flexair suspension at extremes of 1-inch peak-to-peak excursion.

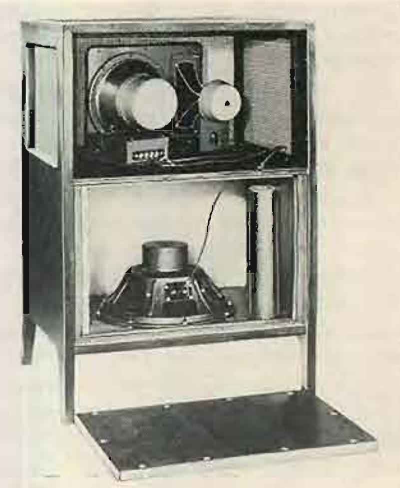


Fig. 12. SS100 3-way System with Stereo Director.

"Q" of 1 would give flat response to 60 cps with the 10-db-down point occurring at 32 cps. The response of the vented enclosure is improved over the range of 60 to 28 cps as compared to the closed box. The transient response and nonlinear distortion of the vented enclosure is, in addition, vastly superior to that of the closed box. In cases where the allowable cabinet volume is more generous, a vented enclosure offers considerable gain in output over the closed box. A stiffness ratio of 2 which corresponds to a volume of about 7 to 8 cubic feet gives a response which is flat down to speaker resonance for a speaker Q of 0.5.

A closed box of similar volume would be 10 db down in output at this point for a similar Q and 6 db down for a Q of 1. The design of this type enclosure is sufficiently difficult that it should be attempted only if adequate test facilities are available. Should the enclosure be mistuned or if a speaker of the wrong "Q" is used, considerable peaking in response can happen at the upper resonant frequency, as shown by *Figs.* 3, 7, 8, and 9. If, however, proper consideration is given to enclosure tuning and speaker damping, the vented cabinet will be the most generally suitable enclosure for medium to small volumes.

New Reproducers and Components

A series of reproducer systems has been designed around the Flexair woofer as shown in *Fig.* 10. These high-compliance, long-travel woofers differ from others now commercially available in four major respects:

1. The low free-air resonant frequency (22 cps for 12-in. P12-NF and 18 cps for 15-in. P15-LF) comes from an increase in compliance of the suspensions rather than from adding mass to the moving systems. Cone weights still are in the normal range, although to the upper side of the conventional tolerance spread.
2. An impregnated cloth bellows-type outer suspension, or annulus, can stretch

greatly with good linearity in force-to-distance ratio. This type of corrugation has been found to be much superior in stretch to conventional "roll-type" corrugations, while still maintaining good lateral centering strength. Polyurethane foam was investigated for the annulus suspension. Its high compliance gave the low resonant frequency required, but linearity became poor long before excursion was adequate. In addition to linearity limitation, air flow through the foam annulus lost considerable bass output efficiency. *Figure* 11 illustrates to exact scale just how the Flexair suspension looks at rest and when stretched to ½" in each direction. The drawing was made from plaster sections cast onto the annulus at rest or stretched. It should be noted especially how the shape is the same for both inward and outward excursion. Edge damping compound stabilizes the annulus movement and eliminates the "edge hole" characteristic of most untreated speakers. This material also effectively seals the interstices of the annulus to prevent air leakage. The coil centering spider is very large, with many corrugations. It contributes only 20 per cent of the total stiffness, and so can use conventional rolls.

3. Efficiency is somewhat higher, largely because low resonant frequency has not been achieved by efficiency-killing high mass. It is within about five to six db of that of conventional stiff supported speakers of the same magnet weight. VU-meter readings taken during listening sessions show power input for generous levels in living rooms seldom rising above three watts.

4. The voice-coil and air-gap relationship is worked out to give a nearly constant drive product over the excursion distance. Air-gap shaping adjusts the lower fringing flux total to more nearly equal that of the upper fringing flux.

Two types of vented enclosure are produced currently by Jensen. Bass-Ultraflex models use rather large rectangular ducts, and the newer, smaller Bass-Superflex models are optimized with stiff, round tubes with outlets very close to the woofer cones.

Reproducers shown here were predicated on furniture styling to current trends and on use of a directional feature for controlling the directional characteristic at higher frequencies to best advantage for actual living-room installations. Extensive testing in homes of Jensen engineers and executives convinced all listeners of the basic merit of physically freeing the upper frequency speaker channels from restrictions forced by enclosure placement. Group audition sessions were held, with listeners recording their preferences in aiming the higher frequency channels as done with the Stereo Director[7,8] assemblies to be seen in the top compartments in *Figs.* 12 and 13.

The direct radiator and horn speaker units in the Stereo Directors have relatively broad directional characteristics, compared to similar units commercially available. It was found that direction preferences were never very exact, and

[7] T.M.
[8] U. S. Patent applied for.

Fig. 13. SS-200 4-way system, with 15-in Flexair woofer and Stereo Director

could not be tied down to within less than 15 deg. However, always there was moderate to strong preference for a general aiming direction. In the great majority of cases, this was in the direction of the listener. Here, however, were found a limited number of contradictory preferences with certain program material types. For instance, with a large symphonic band recording some listeners wanted the Directors aimed outwardly, so as to bounce the sound off walls, or into corners. The explanation given was that this aiming seemed to spread the sound in a manner such as to increase the illusion of a large source.

Conclusions from these tests have been at variance in some respects with certain tests made in treated laboratory rooms. Other conclusions also have not agreed with previously published data. The best explanation offered to date is that no rigid formula can be set up for the way to install a stereo system. Flexibility in movement of speakers, and especially in aiming of the upper channels seems to be quite important at this stage of stereo development.

High back pressure against the cone sometimes is described in terms of "restoring force" or "air spring" for the high compliance woofer in a small closed box. This back pressure cannot be considered a form of loading, however, nor as damping. Damping can only be achieved by electromagnetic coupling to the amplifier or by means of the resistive component of the enclosure impedance.[9] An air volume in a box has no significant resistive component. The vented enclosure, on the other hand, transfers radiation resistance at the port to loading resistance at the cone, thus qualifying as true damping.

The higher internal pressure generated in the tube vented system makes the problem of rigidizing and sealing the woofer enclosure of even more consequence than with the closed box. Using sine wave input well within the power handling capability of the system, easily perceptible air current can be felt up to 12 feet from the mouth of the tuning tube. At 2 feet distance, flame from a "windproof" lighter can be extinguished.

Each woofer is tested with a stethoscope for air leakage between front and back of cone, and each reproducer is tested for air leaks with the woofer mounted and operating at low-frequency sine-wave input. There can be no compromise degree of sealing, if noticeable distortion and output differences are to be avoided. All personnel involved in development and production of the reproducers agree, however, that the extra problems and care are more than offset by the quality and quantity of bass possible in such a small space. Æ

[9] L. L. Beranek, *"Acoustics,"* p. 205.

DAMPING FACTOR CHART

(from page 25)

An experimenter wants to measure the damping factor of an amplifier having an 8-ohm nominal output impedance. For Scale B, he uses a standard 10-per cent 8.2-ohm resistor for R_1, and arranges to switch a 10-per cent 0.82-ohm resistor in series with it for R_2. Still assuming the worst, say that these resistors are off by their full 10 per cent—in opposite directions, of course; R_1 is really 7.38 ohms, and R_2 is 7.38 + 0.902 = 8.282 ohms. This means that our economical experimenter is operating with $R_2 = R_1 \times 1.122$. He goes through the prescribed motions and gets a deflection of, say, −3.2 db.

Consulting a table of db ratios, we see that −3.2 db corresponds to a voltage ratio of 0.6918. If we let $E_1 = 1.0$ volt, for the sake of simplicity, we get the following set of values: $E_1 = 1.0$ volt, $E_2 = 0.6918$ volts, $R_1 = 7.38$ ohms, $R_2 = 8.282$ ohms. The damping factor is, therefore:

$$\frac{7.38}{\left(\dfrac{0.6918-1}{\dfrac{1}{7.38}-\dfrac{0.6918}{8.282}}\right)} = -1.250$$

The damping factor represented on the chart for −3.2 db deflection is −1.205, so the error is about 3.6 per cent.

This error reduces to less than 1.5 per cent if 5-per cent resistors are used, and to less than 0.5 per cent for 1-per cent resistors.

Factor and Fancy

Actually, there is a great deal of fiction in any quantitative statement of damping factor. By its very nature, the term is limited to an expression of the relationship between two supposedly definite and fixed quantities: the internal impedance of the amplifier and the impedance of the load. This is fine as long as the amplifier is terminated by a fixed resistor of proper value—then the term "damping factor" has a real meaning. However, I submit that the whole issue is how well the amplifier can damp *not* a fixed resistor, but a loudspeaker.

An amplifier having an internal impedance of +4 ohms at its 8-ohm tap is said to have a damping factor of +2. But when terminated by a loudspeaker, "+2" accurately represents the measure of control over the load only over a narrow band of frequencies. At some frequency at which the effective impedance of the speaker is 16 ohms instead of 8, the effective control will be twice as great (assuming the internal impedance to remain constant, which it does not and may vary markedly in either direction), and the *effective* damping factor will be +4 instead of +2.

In the days when all amplifiers operated in the positive region, the rated (or nominal) damping factor, therefore, sort of represented the minimum value of control, so that any error was just so much to the better (assuming voice-coil impedance never to fall below nominal). But that doesn't go any more—indeed, it is just the other way around in the negative region: Obviously, if in the foregoing example the internal impedance had been −4 ohms, then the nominal DF becomes −2 and the effective DF becomes −4; twice as bad instead of twice as good.

It occurs to me that there will be times when it will be necessary to state positively the damping characteristics of an amplifier and speaker system combined, which would be in the form of a curve running through the frequency spectrum. I suggest, therefore, that we should adopt the terminology "nominal damping factor" and "effective damping factor" with the added qualifiers "indicated" and "true" where applicable. In most cases, the qualifiers could be omitted, being either understood or unimportant to the purpose at hand. Æ

EQUIPMENT REVIEW

(*from page 46*)

exceptionally acceptable speaker system.

Supplied in 8-ohm impedance only, the X-100 is available in African mahogany, Swedish birch, or pewter walnut. It measures 24 in. high by 24 in. wide and 15½ in. deep, and a cabinet of identical appearance is available for housing phono and amplifier equipment with a designation of X-100-E.

Technically, the X-100 consists of two 12-in. woofers and a single cone tweeter, with adequate cabinet damping for excellent transient response. Efficiency is relatively high for small speaker enclosures, with ¼ watt being sufficient to drive the speaker at full room volume. M-28

Fig. 2. United Speaker Systems' new Model X-100.

STEREO CONSOLE

(*from page 40*)

rectly to the right amplifier, and the left channel goes through the preamp (This gives an opportunity for balancing the two signals, since once the tape recorder playback volume is set, the preamp volume control adjusts the level of the left channel only). We connected everything up, turned on a stereo demonstration tape, and sat back to enjoy ourselves. But it didn't sound "right." "The signals are out-of-phase" we thought, and hastily reversed the speaker leads on one channel. Now the stereo tapes sounded fine, but when we switched back to a monophonic source, *it* didn't sound right. Then the light dawned. Because of the number of stages in this particular preamp, the output signal is 180 deg. out-of-phase with the input signal. This is of no significance in 999 installations out of 1000, but in our case it meant that either the stereo signals would be in phase, or the mononophonic signal, but not both.

Various alternatives were considered, including reversing the phase of the lower playback head in the tape recorder, but the final solution chosen was to alter the output stage of the pre-amp. Since the plate is 180 deg. out of phase from the cathode, a plate-load resistor was installed and the output taken from there instead of from the cathode follower. This put the output signal in phase with the input signal and solved the problem. (We have not, as yet, encountered phasing differences between recorded tapes. If we do it will be necessary to install a reversing switch on one pair of speaker leads. But surely tapes will be standardized as to phase sometime in the future.)

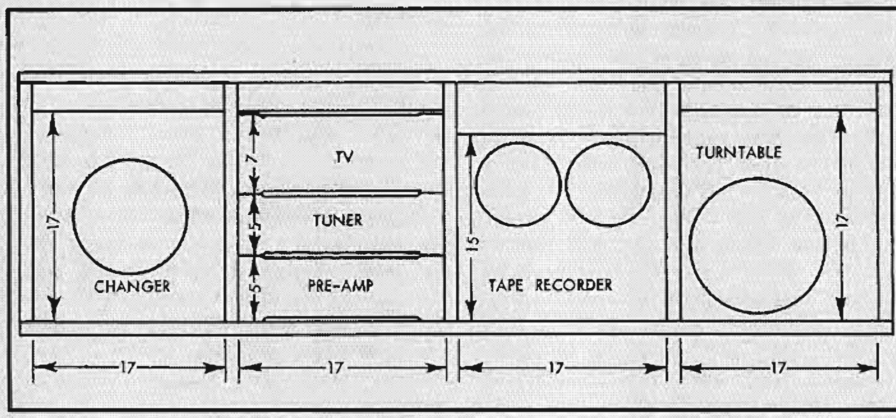


Fig. 5. Plan of equipment panel area. Space is provided for future Conrac Fleetwood TV tuner.

Cabinet Design

The design of the cabinet is conventional except for its size (See *Fig.* 3.). It is a full six feet wide. The speaker enclosures in the cabinet are angled outward which improves the stereo effect with relatively narrow-spaced speakers. A lift-

Fig. 6. Tuner and preamp panels are mounted on individually replaceable mounting boards. Precut panel is for TV tuner.

top lid was chosen since the operation of the record players and the tape recorder are most convenient from the top. The prototype had a full-width lid, but later models use a three-section lid as shown in *Fig. 4*. The tuner and pre-amp are mounted vertically on individual mounting boards with grooved edges. This permits good ventilation and is also an aid in servicing, since each unit may be lifted out individually. In addition, spare blank panels are available so components may be replaced at a later date. We have also included provision for a remote-control television tuner, as shown by the blank cut-out panel in *Fig. 5*.

All in all, we feel we have achieved what we have set out to do—build a high quality wide-range sound system for the home in a single, reasonably handsome, albeit large, console. The basic cabinet shell is adaptable to various components, and less-expensive versions incorporating lower-cost components have been built with some sacrifice in performance.

Future Developments

Well, the next step is, of course, the incorporation of stereo discs. We think the technique of "flat" reproduction on the right stereo channel is a satisfactory one, and the technique we plan to follow is to use an additional tone-arm with stereo cartridge on the turntable and feed the twin high-level equalized signals to dual ganged volume controls on the turntable mounting board. The left signal will feed the AUX position on the Bogen, and a TAPE-DISC selector switch will be added between the outputs of the tape player and the stereo disc reproducer. Operation of the unit will be somewhat more complicated but that is the price we have to pay as the functions are increased.

Please! Someone bring out a stereo preamp with provision for off-the-tape monitoring! It would then be worthwhile redesigning the unit to incorporate a stereophonic tape *recorder* as well as playback. But the beauty of our design, we feel, is that it cannot be obsoleted. As new components and features become available, they can be accommodated in the existing cabinet with ease. Æ

OUTPUT CIRCUIT

(*from page 23*)

transformer turns ratio then should be $24.2/\sqrt{11}$ or 6.7. This calls for a primary impedance of 360 ohms, and transformers of this sort are not presently available. However, a transformer of this turns ratio is not hard to wind by hand; and in the cathode circuit its characteristics are not very critical except for the leakage inductance[4]. An easy way out of the core problem is to buy a cheap transformer of about the right power rating from the local bargain counter and use the core from it. Choose wire sizes so that the secondary resistance is only a fraction of an ohm and the primary resistance proportionally small. The primary and secondary winding should be wound in the closest possible proximity (bifilar, if you have the patience) to keep the leakage inductance low. Primary inductance will be high enough if any reasonable number of turns is put on the transformer; it has only to be about a henry or so for response to 20 cps.

Due to the low gain of triode output stages, a driver stage must be provided. At the input to this stage the signals must be mixed in some way. A resistive mixing bus is one possibility, but it suffers from gain loss and possible degradation of high-frequency response due to its high impedance. An alternative method is suggested by the simplified grid-cathode mixer of *Fig.* 4. Here V_1 and V_2 act as a push-pull voltage amplifier for E_1 except that cathode-follower V_3 also drives their cathodes in a grounded-grid arrangement. The result is addition of E_2 at both sides of the output signal as desired.

In *Fig.* 5 is shown the complete output and driver stages of the "Bi-ortho" circuit. Using the 6BX7 as shown it is capable of about 8 watts output. No phase splitter is shown because nearly any type is satisfactory; channel 1 requires about 2 volts grid-to-grid drive and channel 2 requires about 6 volts.

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[4] Ralph C. Johnston, "A Cathode-Follower Amplifier." *Radio & Television News*, October, 1955, p. 124.

COMING HI FI SHOWS

Nov. 7–9—Omaha, Neb.; Paxton. (*Rigo*)

Nov. 7–9—Detroit, Mich.; Detroit Leland. (*Electronic Representatives, Inc.*)

Nov. 14–16—Cleveland, Ohio; Hotel Carter. (*Cleveland Press, Sta. WDOK, Tri-State Audio Reps. Assn.*)

Nov. 21–23—Seattle, Wash.; New Washington. (*Rigo*)

Nov. 23–30—Mexico City; Fourth Annual Audio Fair. (*AMIAF*)

Jan. 16–18—Minneapolis, Minn.; Dykman. (*Rigo*)

Feb. 5–8—Washington, D. C. International High Fidelity Music Festival of 1959. Shoreham Hotel (*Independent*).

Feb. 6–9—San Francisco, Cal.; Cow Palace. (*IHFM, Northern Cal. Audio Shows, Inc.*)

Feb. 16–23—Los Angeles, Cal.; Biltmore. (*IHFM*)

Mar. 6–8—Denver, Colo.; Cosmopolitan. (*Rigo*)

Mar. 20–22—Baltimore, Md.; Lord Baltimore. (*Rigo*)

Apr. 3–5—Pittsburgh, Pa.; Penn-Sheraton. (*Rigo*)

ELECTROMECHANICAL MATRIXING

(from page 28)

by 45 deg. To do this one must form the sum and the difference of the two driving signals, as in *Fig.* 3.

Positioning of the Driving and Feedback Coils in the Magnetic Field

For distortion-free transduction of electrical energy into mechanical motion, it is desirable to utilize an electro-dynamic system. For a two-motion dynamic feedback system it is necessary to provide one driving and one feedback coil for each motional direction. How such a coil arrangement is placed on a thimble-shaped armature is shown in *Fig.* 4.

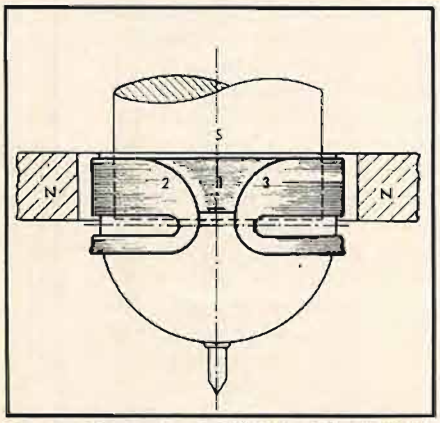

Fig. 4. Diagram of driving-coil arrangement.

The drive coil 1 (circular shape) moves the armature in a ring-shaped magnetic gap in a vertical direction. The two half-coils 2 and 3, on the other hand, are so interconnected, that the conductive parts of those coils which are situated in the same concentric magnetic field as coil 1, produces a rotational motion of the armature. By selecting the point of rotation, this rotating motion imparts to the stylus, for all practical purposes, a lateral excursion. Therefore the motions resulting from coil 1 and a combination of coils 2 and 3, stand at the tip of the stylus at the right angles to each other. The wiring pattern of a 45/45 cutter consists of a further combination of these two coil windings.

The two feedback coils are located near the stylus end of the armature, since the motions at this end are perpendicular to each other and are not, like those at the driving coil point, composed of a vertical and a rotational motion. At the feedback coil location, therefore, a different magnet structure is necessary. The ring-shaped magnetic gap is so constructed as to produce lines of force running at an angle of 45 deg. to the motion of the armature. One can picture these lines of force as being separated into two components, as shown in *Fig.* 5. Vertical motion of the armature permits its ring-shaped winding, 1 to cut the lines of force shown as dashes. This produces a voltage proportional to the velocity of the vertical motion. Lateral motion of the armature produces a cutting of the dotted lines of force, but an emf of equal magnitude and opposite polarity is generated by the two sides of winding 1 therefore producing no output voltage for this motion. The exact opposite takes place in the figure-eight windings 2 and 3. A horizontal motion produces an emf proportional to lateral velocity, while vertical motion produces cancellation in those windings. All this is true provided complete symmetry is maintained. The potentiometer shown in *Fig.* 5 permits balancing of the coils 2 and 3, thereby insuring motions perpendicular to each other. Channel separation in the stylus motion exceeds

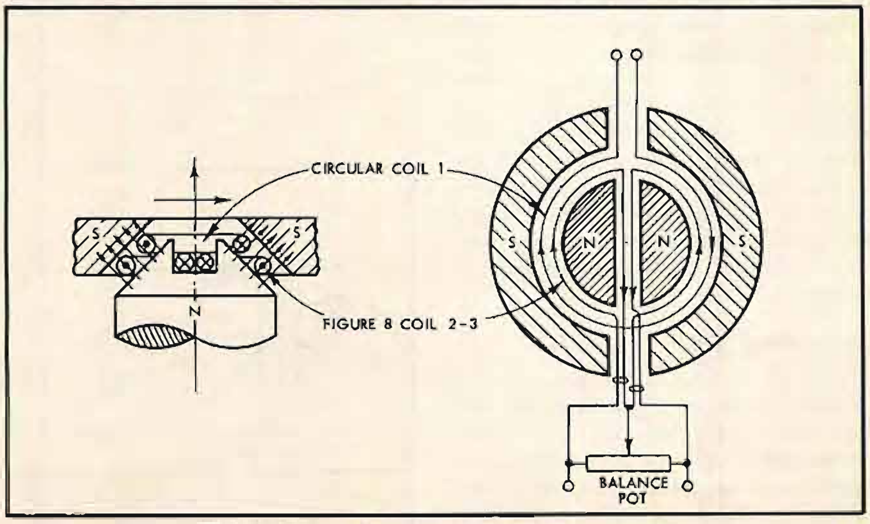

Fig. 5. Feedback coil winding arrangement.

40 db. Should a motional cross of 45 deg. to the record surface be desired, a combination of connections of the feedback coils will also be necessary.

The arrangement just described works with a concentric magnetic system. The ring-windings are always assigned to the vertical motions of the armature, while the bucking half-windings are always assigned to the lateral motions of the armature. This holds true for both the driving coils as well as the feedback coils. Should the concentric placement of magnets be substituted by a transverse one, a reversal of the coil functions will take place. We have chosen the concentric arrangement because of weight considerations. A magnetic field of tremendous strength is necessary (B is greater than 12,000 gauss) and the resulting small dimensions of gap and armature require extreme accuracy in the parts which comprise the magnetic path. It was nevertheless possible to maintain the gross weight of the cutterhead at 12.8 ozs. In a cutter system using a diagonal arrangement of magnets, this weight would be almost impossible to maintain, since the gaps on the left and right must be arranged with alternating *N* and *S* poles, requiring a considerable increase in magnet size.

One of the major difficulties encountered in the design of this double feed-back, two-motion cutterhead was the shielding of the feed-back coil from its driving coil to prevent inductive coupling between them. The close physical location of these coils to each other, the high driving currents and the small order of feedback voltages, made this task very difficult. It is to be noted that the necessary amount of feedback lies somewhere between 40 and 50 db, and that consequently the inductive coupling must be smaller than that. A detailed description of the measures taken to ensure shielding between drive and feedback coils would be too extensive for this article, but the basic difficulty should be noted. Tremendous demands are also made on the heat tolerance of the driving coils and their necessarily tight attachment to the armature. Reason for this are the tremendous acceleration magnitudes (or braking magnitudes) and the driving currents associated with them, especially at high frequencies. Current flow of 200 to 300 amperes per mm^2 under transient conditions are not uncommon.

The Armature and its Suspension

In order to permit the recording of vectoral signals in a single groove, the stylus point must be capable of circular motion. Motion in the direction of groove travel is, on the other hand, not tolerable. The armature therefore must be so suspended as to permit motion in only one plane. In the system developed by Teldec, the thimble-shaped armature is suspended using two parallel leaf springs. These two springs are shaped in such a manner as to place the point of rotation at the center of gravity. The design of these springs was further influenced by the consideration of the resonant frequencies of the springs and the desire that their resonances fall into a purposeful frequency range, as well as close to one another. As already described, the armature is so driven as to produce a vertical motion during which the two leaf springs execute a simple bending motion. When driven to a rotational motion, spring A produces a transverse deflection while spring B executes a like deflection in the opposite direction.

Aside from these two motions, an unwanted vibrating motion is also produced. In this third motion, the springs deflect also in the direction of their maximum stiffness. But even this motion has its resonance placed within the maximum feedback range where it is largely corrected. Decisive in the selection of the resonant frequency were two considerations. For reasons of output, the resonance should be placed at the point where the greatest velocities are to be expected. This point has been determined, on the basis of amplitude statistics, to lie near the center of the usable frequency band. This region is also most favorable to electromotional feedback, since in this way the frequency range to be recorded can most readily be stabilized. Feedback in this cutter is effective in a region from 4½ octaves below to 4½ octaves above the resonant point. Besides the two principal resonances which are caused by the elasticity and the mass of the armature, there are also some secondary resonances. It is at the point of these secondary resonances that the feedback coils are no longer in phase with the drive coils and therefore produce positive rather than negative feedback. To avoid this instability, the first secondary resonance must lie well outside the reproduced frequency range; as a matter of fact it should be removed by at least one octave from the highest reproduced frequency. The secondary resonances can be influenced by a choice of

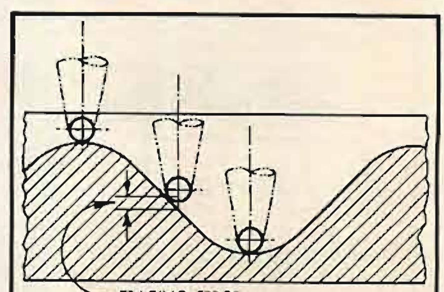

Fig. 6. Section of vertically modulated groove to show how tracing error is caused.

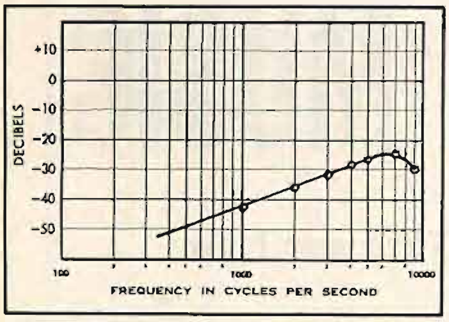

Fig. 7. Distortion in the vertical channel due to pinch effect. Recorded lateral velocity for 0-db reference is 8 cm/sec. Stylus-tip radius, 0.6 mil; high-frequency pre-emphasis, 50 μ-sec. Disc speed, 45 rpm; groove diameter, 12 in.

material and shape for the armature. We chose a material with a relatively large sound transmission coefficient (K = 33,000 ft/sec.). The use of this material placed the first secondary resonance well above 40 kc. For all practical purposes this produces an extremely stable feedback system, almost wholly independent of the stylus dimensions.

The Double-Component Cutting Method

The mechanical recording characteristic of a two-component system differentiates itself sharply from the normal lateral method, in that the ever-present vertical component produces an ever-changing cutting mode of the stylus. Distortion referred to the motion of the stylus itself nevertheless stays extremely low due to the large amount of motional feedback employed. Care must be taken, however, to ensure that no plastic changes (spring-back) occur in the groove. This requires great care in the grinding of the stylus itself and in the selection of lacquers for mastering. The stylus burnish must be selected as the optimum for the mechanical properties of the acetate material used, and the temperature of the heated sapphire becomes more critical.

If we now play this recording with a reproducing sapphire which has a ball-shaped tip, we meet up with further causes of distortion, some of which are not found in lateral recording. The main

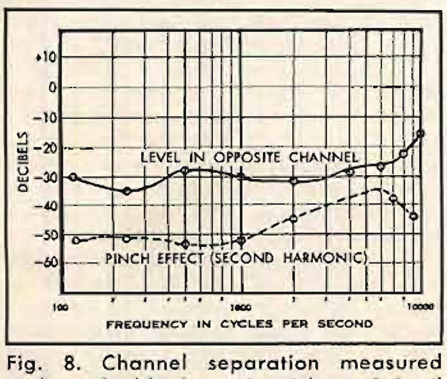

Fig. 8. Channel separation measured with a double-dynamic pickup, 0.6-mil tip radius. Recorded velocity, 2.5 cm/sec. for 0-db reference.

cause of this distortion is the pinch effect, which produces second harmonic distortion in the vertical channel caused by excursions in the lateral direction. A second distortion-producing problem results from the vertically recorded component. Since the reproducing stylus has a finite size, an exact tracing of the vertical groove component is impossible (*Fig.* 6). Here as well, the second harmonic is produced with effects in the same direction as the excursion, i.e. in the vertical direction.

These distortion figures are symmetrical only if the modulation-cross is inclined at an angle of 45 deg. to the record surface. The amplitudes of the vertical component then assume a magnitude in each of the 45-deg. directions of only 0.7 or a reduction of 3 db. In a vertical-lateral system, on the other hand, only the vertical channel is adversely affected by this distortion although in the full measure. If a reasonable relationship is maintained in the dimensions of the groove, recorded level, groove velocity, and reproducing stylus, these interferences can be held to a minimum, and reproduction can attain the demands which are made in today's state of the electroacoustical art. *Figure* 7 shows a plot of the distortion imparted to the vertical component by the pinch effect as related to frequency. The curve reaches its maximum around 7.5 kc. The reason for this is the dropping playback equalization curve (RIAA) which tends to suppress this form of distortion at higher frequencies. Furthermore, the second harmonic of 7.5 kc lies already at 15 kc where reproduction falls off rapidly in most systems. These data were obtained using a Neumann "DST" double-dynamic reproducer. *Figure* 8 shows the channel separation. Here again we can see the typical interference of the second harmonic caused by the pinch effect.

These quality limitations, when examined in the light of the above factors, produce the limit of the innermost groove diameter.

Conclusion

The transducers described here were developed in the Laboratories of TELDEC (Telefunken-Decca) in Berlin, Germany. They have been in use for over a year now for the purpose of providing practical knowledge in the cutting of stereophonic disks. As a result of this considerable experience it has been proven that stereophonic recordings may be transferred to disc in such a manner as to make differentiation between the original tape and the disc impossible even for trained ears.

We wish especially to acknowledge the contributions of Georg Neuman Laboratories, Berlin, whose practical construction of the cutting head have made our developments possible. Æ

ABOUT MUSIC

The Conductor—Then and Now

HAROLD LAWRENCE*

The baton, a light wooden stick with a cork handle, is the familiar symbol of the conductor's authority. Nowhere is this authority more dramatically underlined than at the start of a symphonic concert, when the house lights dim and a hush settles over the hall as the conductor raises his baton. In that moment, the audience's attention is focused on the silhouetted figure on the podium while ninety players watch the tip of the baton for the downbeat. In the hands of a master conductor, the short white "rapier" can be a supple and expressive instrument. But it was not long ago that the baton was merely a time-beater and the conductor an unobtrusive musical overseer.

Conducting as we know it today did not exist before the 19th century, although there were, of course, methods of controlling musical performances. Who then were the forerunners of the modern conductor, and what techniques were employed to direct an orchestra?

The late 17th- and 18th-century "conductor" was first and foremost a composer, whose primary function was to write music for his patron. He was proficient in one or more instruments (usually the keyboard and violin) and participated in performances of his own works. He was also expected to supervise these performances, and in this he had some help.

The composer-conductor did not ascend a podium, as does his modern counterpart. His was a more humble position. Seated at a keyboard instrument (harpsichord or organ), he struck chords from a figured bass, occasionally stepped in with a firm beat when the ensemble floundered, saw to it that the parts had been correctly copied, and acted as a sort of concert and personnel manager besides. The task of "conducting" the instrumentalists was assigned to the first violinist, or concertmaster. Together the keyboard-conductor and concertmaster directed the orchestra, though their roles differed in importance according to the work involved.

In purely instrumental music, the concertmaster was the principal conductor, while the keyboard-conductor was secondary. In opera, where the stage took precedence over the pit, the keyboard-conductor was the guiding force, accompanying recitatives, watching over the vocal performance, and superintending the chorus. The orchestral playing, however, was still left in the hands of the concertmaster.

* *26 W. Ninth St., New York 11, N. Y.*

Early Hazards

In Paris, a *batteur de mesure* (time-beater) assisted the keyboard-conductor. Wielding a baton the size of a herald's staff, he pounded the beat on the floor. (Lully, incidentally, was said to have died a victim of his own time-beating; during a rehearsal at the *Opéra,* he accidentally mashed his toe with the big stick, and a subsequent infection brought about his death.) This noisy time-beating, which was not confined to rehearsals, drew caustic attacks from composers and critics. In his *Dictionnaire de Musique* (1756), Rousseau described how Parisian audiences were "shocked" over the "disagreeable and continuous" racket made by the time-beater, whose hammering was so loud that it carried above the sound of the music. It was not until the mid-19th century that audible time-beating was entirely abolished. As late as 1847, François Habaneck beat time with his baton by rapping it sharply against the top of the unfortunate prompter's box.

The division of labor between keyboard-conductor and concertmaster worked fairly well until the end of the 18th century when orchestral writing became more complex. The period of transition from the dual control system to silent, interpretative baton conducting lasted approximately half a century, and was marked by a struggle for power between the keyboard-conductor and the concertmaster.

Individualism

In England, for example, whose musical development traditionally lagged behind that of the Continent, dual control was given up slowly and reluctantly. In 1830 an English musician wrote in *Harmonicon* that "there can be no question of the superiority of the [baton conductor] plan over that pursued with us. For [in the former instance] our ears are not excruciated by two distinct beats, intended to represent the same time: one being clapped by the hands of the *conductor,* and the other being stamped, sometimes furiously, with the foot of the first violinist or leader." It was not unusual to find conductor and leader in total disagreement over the tempo. This was especially evident at such junctures as an *accelerando, ritardando,* or other tempo alteration, when one part of the orchestra went one way with the keyboard-conductor and the rest with the leader. Managing of dynamics presented another thorny problem, and every once in a while one conductor would hush the orchestra while his colleague called for a *fortissimo*.

Another drawback of the dual control system was the fact that conductors were bound to their respective instruments. Besides playing his part, the concertmaster also beat time for the orchestra by waving his bow, rapping his music stand, wagging his head, stamping his foot, and even bouncing in his chair. His own violin technique invariably suffered as a result, wrote a 19th century organist, "[of] the supposed necessity of making the fiddle growl, grumble, shout, scold, and admonish the arrant propensities of other instruments." Control of the orchestra from the first violin desk was also hampered by that fact that the concertmaster's orchestral "score" (at best a two-staff reduction) gave him only a sketchy idea of the orchestration.

The keyboard conductor, on the other hand, generally had the full orchestral score before him, but was less strategically placed than the concertmaster to shape the instrumental playing. And since there was no longer any need for him to "fill in" harmonically, the piano became not only a superfluous but an intrusive tonal color.

As the orchestra grew in scope and proportions, keyboard-conductors and concertmasters began to play less and conduct more. Some took the bold step of breaking their instrumental ties altogether in order to devote themselves entirely to conducting. In France, where the concertmaster had always played a dominate role, most of the "new" conductors were ex-violinists who used their bows as batons. In Germany, the keyboard-conductor-composer generally took over as sole baton-conductor. England, on the other hand, might have retained her outmoded dual control were it not for the fact that prominent musical visitors like Spohr and Mendelssohn proved baton-conducting superior.

Early Batons

Before the modern baton was established as an international standard, a wide assortment of implements ranging from a roll of paper, a thick staff resembling a drum-majorette's baton with ivory knobs at each end, a roll of leather stuffed with calf's hair, to fiddlesticks, was seen in the concert hall. Nowadays many conductors prefer to to dispense with batons entirely. But it really doesn't matter how a conductor achieves his goal, as Nicolai Malko wrote in his book, *The Conductor and his Baton* (Wilhelm Hansen, Copenhagen):

"Once after a concert Nikisch was in a restaurant where a band was playing. Someone jokingly suggested that he conduct the band. Nikisch approached them, selected a rose from the bouquet on the table and, grasping it by its stem, began to conduct the *Merry Widow* Waltz. At times he almost touched the first trumpet soloist in the face with the flower. Probably the band never played better than it did that night." Æ

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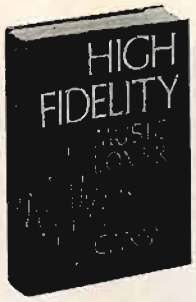

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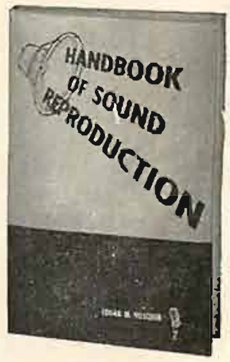

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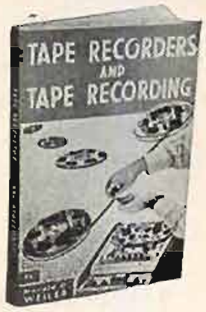

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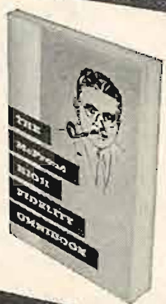

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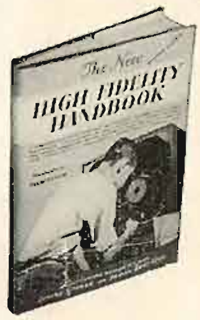

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DB, DBM & THE VU

(from page 32)

differs from the db in the value of the reference level, in application, and in the type of meter required for measurement purposes. Although it is possible to use decibal notation with any waveshape, the VU is restricted to the measurement of complex audio waves. Furthermore, any reasonably good meter having flat frequency response out to about 20 kc may be calibrated for db readings, but VU meters must be built according to exacting specifications. The final difference between the units is of course the reference level, the VU being based on a 1 milliwatt power dissipation across 600 ohms as its zero value. Equation (*1*) may be used for VU calculations by substituting the appropriate reference level.

Almost everyone relies on a sinusoidal oscillator for testing purposes. Consider for a moment the problem involved in a broadcast studio if the sine wave generator output is measured in db, and the audio circuits are monitored with VU meters. It would be necessary to add and subtract values for each reading in order to arrive at any conclusions, for the two units operate with different references. Thus, zero on a db meter is not zero on a VU meter, and *vice-versa.* Clearly, something is needed to eliminate extra calculations, and a db unit based on the same reference level as the VU would be quite handy. To prevent confusion between the standard decibel with its 6-mw reference and the new unit, using a 1-mw reference, the latter is known as the *dbm,* signifying decibels with a 1-milliwatt base. Use of the dbm eliminates the necessity of converting from one unit to another when making measurements with sinusoidal generators. As a further aid in simplifying measurements, most modern db meters are in reality dbm meters, that is, the zero marking on the scale corresponds to 1 mw/600 ω rather than 6 mw/500 ω.

To further illustrate the application of the various reference levels, consider the following examples:

Example 1. Find the power level in db for a circuit in which the power across the load is 5 watts.

Solution: Use Eq. (*1*) and substitute. Since the desired value is to be in decibels, the 6-mw reference is used. Thus

$$N_{db} = 10 \log (5/.006) = 10 \log 833 = 29.21 \text{ db}$$

Example 2. Find the power level in dbm for the power specified in Example 1.

Solution: Again, use Eq. (*1*), but in this instance the reference level is 1 mw.

$$N_{dbm} = 10 \log (5/.001) = 10 \log 5000 = 36.99 \text{ dbm}$$

Example 3. The signal on a telephone line is +4 VU. How much power (in watts) is on the line?

Solution: Transpose Eq. (*1*), and solve for P_1

$$P_1 = P_2 \log^{-1} (N_{vu}/10) = .001 \log^{-1} (4/10) = 2.51 \text{ mw.}$$

Example 4. What is the level in dbm corresponding to 0.3 mw of power?

Solution: Use Eq. (*1*) with the 1-mw reference level.

$$N_{dbm} = 10 \log (.0003/.001) = 10 \log (0.3) = 10 \; (9.477 - 10)$$

As discussed previously, the negative characteristic makes this calculation awkward. However, by subtracting 19 from 9.477 the work is simplified:

$$N_{dbm} = 10 \; (-.523) = -5.23 \text{ dbm}$$

Voltage Reference Levels

Equation (*1*) permits the calculation of db, dbm, and VU levels only when the power content of a signal is known. Measuring power in an amplifier circuit is usually difficult, and involves opening the circuit at some point. As a practical matter, therefore, measurements in most circuits are invariably made with a voltmeter, since no connections need be broken or altered. It seems apparent that an equation similar to Eq. (*1*), but based on a voltage ratio, would be helpful. In order to derive such an equation, it is only necessary to recall that power in an electrical circuit is found from the expression E^2/R. Substituting this into Eq. (*1*) for both P_1 and P_2 gives

$$N_{db} = 10 \log \frac{E_1^2/R_1}{E_2^2/R_2} = 10 \log \frac{E_1^2}{E_2} \times \frac{R_2}{R_1}$$

But, the logarithm of a product is equal to the sum of the individual logarithms, and the logarithm of a squared term is twice the logarithm of the term in linear form. This permits the above equation to be rewritten as

$$N_{db} = 20 \log \frac{E_1}{E_2} + 10 \log \frac{R_2}{R_1} \qquad (2)$$

in which E_1 is the rms voltage of the signal being measured, E_2 is the rms reference voltage, R_1 is the resistance across which E_1 is measured, and R_2 the appropriate reference resistance obtained from the power reference (either 500 ω or 600 ω).

In order to utilize Eq. (*2*), it becomes necessary to express the reference level in volts. Using the same expression for power ($P=E^2/R$) and solving for E shows that

$$E=\sqrt{PR}$$

Then, substituting the proper values of P and R from the power reference levels, for a 1-mw reference

$$E_2=\sqrt{(.001)(600)}=0.775 \text{ volts}$$

Similarly, for a 6-mw reference, $E_2 = 1.732$ volts.

By substituting I^2R for P in Eq. (*1*), another useful db relationship may be formed. Thus, for current,

$$N_{db}=20\log\left(\frac{I_1}{I_2}\right)+10\log\left(\frac{R_1}{R_2}\right) \qquad (3)$$

Inasmuch as it is simpler to measure voltage in a circuit, a current reference for I_2 is not normally used, although one could be calculated. Equation (*3*) finds its application in circuit analysis when currents are being calculated rather than measured. I_2 is considered to be a reference only for the circuit under discussion at the moment, and may have any value whatsoever. R_1 and R_2 in this instance are the resistances through which I_1 and I_2 flow.

Correction Factor

The term $10\log(R_2/R_1)$ which appears in Eq. (*2*) is known as a correction factor. Its purpose is to raise or lower N_{db} when the resistance across which E_1 is being measured is not equal to the appropriate reference resistance, R_2. The correction factor corrects N_{db} for the *numerical* difference between voltage and power in the circuit. Thus, if E_1 is measured across the proper value of resistance (500 ω or 600 ω) the correction factor may be neglected since it becomes $10\log 1$, which, of course, equals 0. However, if R_1 is greater than the reference resistance, the correction factor will be negative, and N_{db} will be reduced. Conversely, if R_1 is less than the reference value, the correction factor will cause N_{db} to increase.

In certain applications the correction factor is omitted when using Eq. (*2*). For these cases, the resulting value of N_{db} is known as the *apparent level*, so named because meters calibrated for db readings only respond to the voltage being measured and completely neglect the resistance across which measurement is made. On the other hand, Eq. (*1*) always gives the correct level, regardless of resistance, so the values calculated from this equation are known as *true levels*. When the correction factor is equal to zero, the apparent level and the true level are equal. For all other cases, a simple relationship exists:

$$N_{db_{true}}=N_{db_{app}}+\text{Cor. Factor} \qquad (4)$$

As an illustration of the need for the correction factor, consider the circuit of

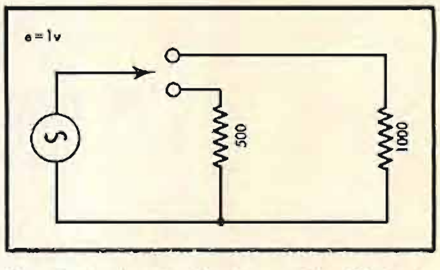

Fig. 5. A simple circuit used to illustrate the need for the correction factor in db calculations.

Fig. 5. The switch permits the use of either a 500-ω or 1000-ω load. For simplicity, the generator has no internal impedance, and provides 1 volt across the load regardless of switch position. When the switch is placed in the 500-ω position, the load power will be

$$P=E^2/R=1/500=2\ mw.$$

Now, using Eq. (*1*) to find the true level in db:

$$N_{db}=20\log(1/1.732)=-4.77\ db$$

Using Eq. (*2*) (but without the correction factor) to find the apparent level:

$$N_{db}=20\log(1/1.732)=-4.77\ db$$

The two calculations provide identical results only because the resistance of the load was equal to the reference resistance. However, if the switch is thrown to the 1000-ω position, the load power becomes

$$P=1/1000=1\ mw.$$

and the true level, from Eq. (*1*) is

$$N_{db}=10\log(.001/.006)=-7.78\ db$$

The apparent level, from the voltage ratio of Eq. (*2*) is

$$N_{db}=10\log 1/1.732=-4.77\ db$$

Observe that the apparent level is higher than the true level in this illustration since the correction factor is not zero. The correction factor may be found, and added to the apparent level, in which case

$$N_{db_{cf}}=10\log R_2/R_1=10\log(500/1000)=-3.01\ db$$

and $N_{db_{true}}=-4.77-3.01=-7.78$ db.

Gain and Loss

In audio circuits, the comparison between input and output levels presents several possibilities. Either the output will be greater than the input, as with amplifiers, or the output will be less than input, as happens with pads, filters, and equalizers. In some circuits (the cathode follower is an excellent example) it is possible that the output voltage will be smaller than the input voltage, while the output power is larger than the input. If these conditions are to be discussed intelligently, it is essential that the circuit parameter involved be specified. As a further aid in clarifying what might become a muddled situation, the IRE has standardized the terms *gain* and *loss* for use with power only. Thus, any system in which the output power exceeds the input power is said to exhibit a gain, while circuits having higher input power levels produce a loss. When voltage is the criterion, a circuit producing a larger output than input is said to provide *amplification*, and if the output is the smaller, the circuit produces *attenuation*. Unfortunately, the standardization of terms is a slow process, and it is entirely possible that the term attenuation may be used when loss is intended. This is particularly true in older textbooks and magazines (prior to 1956). In most instances, however, the text itself will provide a guide to what is meant.

Gain and loss are both pure numbers indicating a ratio of input to output power for a circuit. For instance, if an amplifier provides an output power 1000 times greater than the input, the gain of the amplifier is 1000. This can be expressed conveniently in db, for the db gain of a circuit is defined as the difference in db levels between input and output. Thus

$$N_{db_{gain}} = 10 \log \left(\frac{P_{out}}{P_2}\right) - 10 \log \left(\frac{P_{in}}{P_2}\right)$$

which, by the rules of logarithms, may be written as

$$N_{db_{gain}} = 10 \log \left[\frac{P_{out}/P_2}{P_{in}/P_2}\right] = 10 \log \left(\frac{P_{out}}{P_{in}}\right) \qquad (5)$$

Note that the reference levels cancel, so that gain (or loss) is always expressed in db, never in dbm or VU.

Fig. 6. A typical audio voltmeter. Although the scale is marked "Decibels," the reference level indicated on the meter is given as 1 mw/600 ω. Observe that 0 on the db scale is directly under 0.775 volts on the rms voltage scale. The meter is in reality a dbm meter. (Courtesy Hewlett-Parkard Co.)

Returning to the example above, if the ratio of P_{out} to P_{in} is 1000, then $N_{db_{gain}}$ is 30 db. In the event the output power is less than the input, the gain becomes negative, indicating a loss. Loss is usually considered a positive quantity, so, when the output power is less than the input, the ratio of P_{out} to P_{in} in Eq. (5) is usually inverted. To illustrate further, a circuit in which the output is 20 db less than the input has a gain, by Eq. (5), of −20 db, or more conveniently, a loss of +20 db.

Although gain and loss are only expressed in db, they may be added to levels specified in db, dbm, or VU. As an example, consider an amplifier having a gain of 50 db. Assuming the input to the amplifier is −40 VU, the output level would be the input level plus the gain, or −40 + 50 = +10 VU. The reference used for the input level is also used for the output level, thereby simplifying the arithmetic involved in gain and loss calculations. Based on Eq. (5), the following relationships may be used for gain and loss calculations:

$$\text{Output level} = \text{input level} \pm \text{gain or loss in db} \qquad (6)$$

$$\text{Input level} = \text{output level} \mp \text{gain or loss in db} \qquad (7)$$

$$\text{Gain} = \text{output level} - \text{input level} \qquad (8)$$

$$\text{Loss} = \text{input level} - \text{output level} \qquad (9)$$

In Eqs. (6) and (7), the topmost sign operates with gain, the bottom with loss.

As with reference levels, it is more convenient to work with voltage rather than power. For gain and loss calculations, however, the input and output impedances of a circuit may not be equal, and may not be purely resistive, requiring that phase angles be taken into account. By substitution, Eq. (5) may be rewritten in terms of voltage, impedance, and phase angle:

$$N_{db_{gain}} = 10 \log \left[\frac{\dfrac{E_{out}^2 \cos \theta_{out}}{Z_{out}}}{\dfrac{E_{in} \cos \theta_{in}}{Z_{in}}}\right]$$

which can also be expressed as

$$N_{db_{gain}} = 20 \log \left(\frac{E_{out}}{E_{in}}\right) + 10 \log \left(\frac{Z_{in}}{Z_{out}}\right) + 10 \log \left(\frac{\cos \theta_{out}}{\cos \theta_{in}}\right) \qquad (10)$$

In Eq. (10), Z_{in} and Z_{out} are the magnitudes of the input and output impedances. The phase angles, θ_{in} and θ_{out}, are the angles whose tangents are X/R.

Amplification and Attenuation

Many circuits, preamplifiers in particular, provide no appreciable power output. It is therefore more convenient to consider input and output voltages only, neglecting the impedances and phase angles that may be present. When voltage is the criterion, the amplification provided by a circuit may be found from the voltage ratio of Eq. (*10*) by itself. Attenuation, on the other hand, is obtained from the same ratio by merely inverting the fraction. Thus:

$$Amplification = 20 \log (E_{out}/E_{in}) \quad (11)$$

$$Attenuation = 20 \log (E_{in}/E_{out}) \quad (12)$$

As with loss, attenuation is assumed to be a positive quantity indicating an output lower in value than the input. Negative values of amplification are actually values of attenuation.

Both attenuation and amplification are based on a logarithmic ratio, and are therefore usually measured in decibels. Originally, however, the decibel was defined in terms of a power ratio. Because power does not enter into calculations of attenuation and amplification, the IRE has suggested that these two quantities be measured in a new unit, called *decilog*, to differentiate them from gain and loss. This seems entirely feasible, and more than likely, newer texts and articles will incorporate this term. The decilog, it should be noted, is merely another name for 10 times the logarithm of a quantity.

The relationships existing between the input and output of a circuit can best be shown by a numerical example. Suppose that a cathode-follower circuit operates with an input of .02 watts across a 10,000-ohm resistance, and provides an output of 0.1 watts across 100 ohms. What is the gain and what is the amplification provided by the circuit? The gain is found from Eq. (*5*), by inserting the input and output power values:

$$N_{db_{gain}} = 10 \log (.1/.02) = 6.99 \ db$$

The amplification may be found from Eq. (*11*), but first the input and output voltages must be found:

$$E_{in} = \sqrt{(.02)(10{,}000)} = 14.14 \ volts$$

$$E_{out} = \sqrt{(0.1)(100)} = 3.16 \ volts$$

and, from Eq. (*11*)

$$Amplification = 20 \log (3.16/14.14) = -13.02 \ db$$

which is therefore attenuation rather than amplification.

Conclusion

It has been shown that the decibel is a unit with which signals can be accurately measured and calculated. Furthermore, the db is a compressed unit which operates on a scale similar to human hearing. Two other quantities, the dbm and the VU, are used in audio circuits for measurements of levels. The db is also used for gain and loss measurements, while a new unit, the decilog, is used for amplification and attenuation. It is the author's hope that the material presented above has been useful and helpful in understanding the decibel, for it is the basic yardstick of measurement in audio circuits. Æ

CLASSIFIED

Industry People...

NOTES FROM THE NEW YORK HIGH FIDELITY SHOW. Elliot Davis, director of shows for the IHFM, chose the New York show to announce his resignation. He is joining **Sid Weiss's** factory representative firm in Los Angeles as a partner . . . **Sid Cottin,** formerly an executive with Golden Crest records, is the new show director for the IHFM . . . **Bill Thomas** and **Ray Pepe,** president and sales manager, respectively, of James B. Lansing Sound, Inc., both pleased and astounded by interest in the company's super stereo speaker system—the Ranger Paragon—which sells for almost two thousand dollars . . . **Emanuel (Bert) Berlant** equally pleased over acceptance accorded the Stephens "Stereo Dot" speaker system.

John Hilliard, chief engineer for Altec Lansing Corporation, and **Harry F. Olson,** who heads up acoustical development for RCA, visited the High Fidelity Show in the New York Trade Show Building only briefly, spending the bulk of their time participating in the annual convention of the Audio Engineering Society, which was held co-incidental with and directly across the street from the Hi-Fi show, in the Hotel New Yorker . . . The Windy City contingent at the Manhattan show included **Bill Grommes,** president of Precision Electronics, Inc., which manufactures high-quality amplifiers under his name; **Karl Kramer,** engineering executive with Jensen Manufacturing Company; **Irving Rossman,** president of Pentron Corporation; and **Maurice Meshboum,** prominent commercial artist who is the dean of Chicago hi-fi enthusiasts . . . **Edgar Villchur** elated over public reaction to the new AR tweeter, first described in the October issue of AUDIO.

Paul Sampson, executive with Harvey Radio Company audio department, accepting congratulations on his October 18 marriage to **Janet Buffington** . . . **Hermon Hosmer Scott,** who suffered a near-fatal illness only recently, surprised friends by showing up looking as fit-as-a-fiddle . . . **James Grayson,** president, Westminster Recording Company, Inc., was one of a number of top executives who attended their firm's exhibits to get a first-hand opinion on public thinking. Others included **Avery Fisher,** president of Fisher Radio Corporation, **George Silber,** president of Rek-O-Kut Company, and **Milton Thalberg,** president of Audiogersh Corporation . . . **Jack Frazier,** president of International Electronics Corporation, Dallas, Tex., represented the second-largest (Ed. Don't fire me, I quit) state with an impressive display of high-quality speakers manufactured under his name . . . **Vincent J. Skee,** head of Electronic Applications, displayed a German-made reverberation chamber at the AES convention which may well alter recording techniques throughout the world. More about this in a subsequent issue of AUDIO. **Lester H. Bogen** has resigned as president of the Bogen-Presto division of the Siegler Corporation . . . **F. S. Yarbrough,** one of the West Coast pioneers in the electronics industry, passed away from a heart ailment on September 24. Until 1955 he was owner of American Microphone Company, which has since been sold to Elgin National Watch Co.

Industry Notes...

AES ELECTS OFFICERS. Donald J. Plunkett of **Capitol Records has** been elected president of the **Audio Engineering Society** for the coming year, succeeding Sherman M. Fairchild. The election was held in conjunction with the Society's recent convention in New York. Executive vice-president is Harry L. Bryant of Radio Recorders, Los Angeles. Other new officers are: Arthur G. Evans, RCA, Indianapolis, central vice-president, and Vincent Salmon, Stanford Research Institute, Menlo Park, Calif., western vice-president. C. J. LeBel, president of Audio Instrument Company, and Ralph A. Schlegel of RKO Teleradio Pictures, both New York, were re-elected secretary and treasurer, respectively. Newly-elected governors are Frank E. Pontius of Westrex Corporation, Ross H. Snyder of Ampex Corporation, and S. Edward Sorensen of Columbia Records, Inc.

ADVERTISING INDEX

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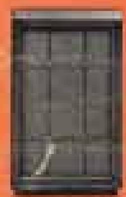

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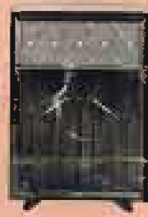

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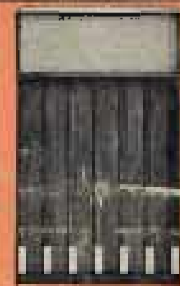

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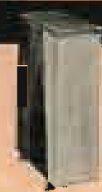

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