AUDIO

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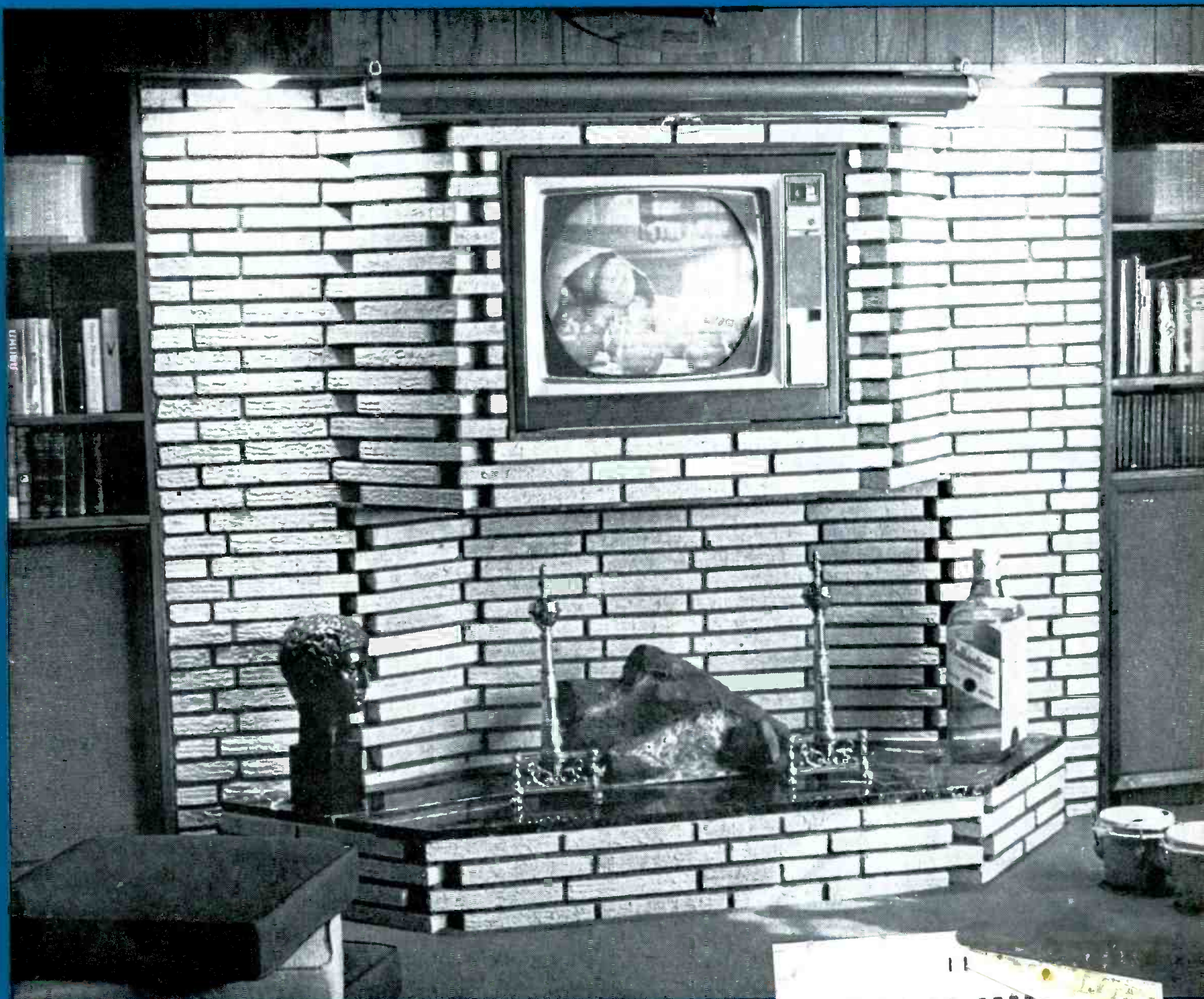
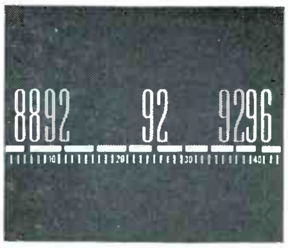
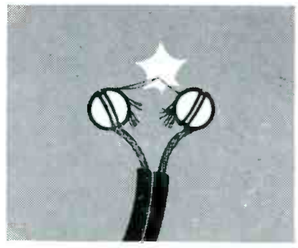


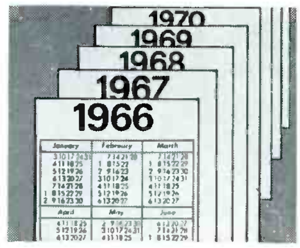
Only Scott can put these exclusive features in a receiver



No chance of signals from strong local stations popping up where they don't belong on the dial and blotting out the more distant stations you want to hear! The 342 incorporates revolutionary new field effect transistor circuitry for maximum tuner sensitivity with virtually no cross-modulation, no drift, no more problems caused by changing tube characteristics. Scott is the first, and only, manufacturer to use this important advance in solid state design.



You can forget about shorted connections burning out expensive transistors because Scott engineers did not! The 342 is designed to withstand these common problems: accidental shorting of speaker terminals, subjecting the input to a high level transient signal, or operating the amplifier section without a load. Special quick-acting fuses protect both your expensive speakers and the transistors themselves.



Scott uses silicon power transistors in the 342. Silicon is more costly than germanium, but far superior in terms of ruggedness, reliability, and resistance to overload, heat and aging. The silicon transistors in the 342 output circuitry provide instantaneous power for even the most extreme music dynamics. This Scott circuit achieves extremely low distortion at low power levels . . . makes all your listening so much more enjoyable.

And Scott puts them all in a receiver under $300

The new 65-watt Scott 342 solid state FM stereo receiver gives you the features, the quality, the reliability, the magnificent sound you've come to expect from Scott . . . and at a price less than $300!

Costing less than ordinary vacuum tube equipment, this no-compromise solid state unit incorporates the popular features of the most expensive Scott components .

Output and driver transformers, major sources of distortion and diminished power, are eliminated from Scott's radically new solid-state amplifier design. As a direct result of transformerless output design, the Scott 342 has high frequency response superior to separate stereo components costing far more.

The Scott 342 includes these important features found in the most expensive Scott components: tape monitor switching, speaker switching with provision for remote speaker selection, switched front panel stereo headphone output, front panel stereo balance switch, individual clutched bass, treble, and volume controls for each channel, fully automatic stereo switching and many more.

THE 342 FM STEREO RECEIVER . . . Usable Sensitivity, 2.7 μv; Harmonic Distortion, 0.8%; Drift, 0.02%; Frequency Response, 18-25,000 cps ±1 db; Music Power Rating per channel, 32½ watts; Cross Modulation Rejection, 85 db; Stereo Separation, 35 db; Capture Ratio, 6.0 db; Selectivity, 40 db.

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AUDIO

January, 1966 Vol. 50, No. 1

Successor to RADIO, Est. 1917

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COMING

Articles

Test Equipment—What you should have to perform routine performance checks on your own equipment and installation.

A Test Tape for 4-track stereo recorder head position adjustment.

De-equalizing circuitry to permit quick checking of amplifier equalizer circuits.

and

Part 2 of Norman H. Crowhurst's "Audio Measurements Course."

Profiles

Heathkit Multiplex Generator

Revox G-36 Tape Recorder

Electro-Voice 1177 Solid-State Receiver

In the February Issue

On the newsstands, at your favorite audio dealer's, or in your own mailbox.

AUDIO CLINIC

Joseph Giovanelli

Send questions to:
Joseph Giovanelli
2819 Newkirk Ave.
Brooklyn, N. Y.
Include stamped, self-addressed envelope.

Compatibility

Q. Can transistorized hi-fi components be used in the same system with tube-type components? The literature I have read on solid-state equipment does not make this point clear.

For example, can a transistorized preamplifier be fed into a tube-type basic amplifier, or a transistorized tape recorder playback preamplifier be fed into a tube-type preamplifier of the high-fidelity system?

If you develop this question in your column in AUDIO, *I would like you to discuss what determines compatibility between solid-state and tube-type components. If they are not compatible, how do you make them compatible?*

I understand that these tube type components are essentially voltage-operated devices and transistorized equipment works on varying currents. If this so, why do conventional earphones and loudspeakers work with solid-state equipment? Carl Stoffels, Chicago, Illinois.

A. If we have a solid-state preamplifier, there is no real reason why we cannot feed this device into any tube-type power amplifier. The main consideration is the impedance of the solid-state device and the impedance of the tube device into which it feeds. So long as the amplifier's input impedance is not lower than the specified load impedance of the solid-state preamplifier, there will not be any difficulty. Because a solid-state device is usually a low-impedance device and because in tube-type amplifiers, input impedance is usually quite high, this criterion is met. The point to remember is that if the amplifier's input impedance is too low, the coupling capacitor in the preamplifier may be too small to pass all the low frequencies into the power amplifier. Therefore, some loss of low-frequency response will be the result.

This same point holds true for a transistorized tape playback preamp. The input impedance of a tube-type preamp is likely to be much higher than required for the output circuit of the tape playback preamp—including the size of the coupling capacitor.

There is sometimes a problem when the device feeding a preamp is a tube-type device and the preamp or power amplifier is a solid-state device. Solid-state devices often have very low input impedances. This fact must be taken into account when selecting driving sources for them. If the impedances are correct as described, you will have no compatibility problem, any more than is true of tube-operated devices connected to other tube-type devices.

Naturally, you must take into account the sensitivity of the power amplifier versus the requirements of the solid-state preamp, just as you do when pairing sets of tube-type components.

You are correct in stating that tube devices are voltage-operated whereas solid-state equipment is current-operated. The concept that stages are voltage-operated or current-operated is more a matter of convenience in discussion than of strict accuracy. The grid circuit of a tube is very high in impedance. However, there is a grid return resistor across which the input signal is developed. As is true of any resistor, some current will flow in it when a voltage drop is present. Therefore, we can say, that to whatever degree the grid resistor draws current, the devices under discussion are current-operated.

The circuits of solid-state equipment are basically low-impedance circuits. Because considerably more current will flow in them than in vacuum-tube circuits, we say they are current-operated devices. However, the resistance of such circuits, even when low, will, because of the flow of current in them, have a voltage drop across them. Thus, to some extent we can say that these devices are voltage operated. Naturally in tube-type devices, the voltage element is most dominant and the current too small to be considered for most applications. The converse is true of solid-state devices. Whether stages are essentially voltage-operated or current-operated devices, they have one thing in common. In a voltage-operated device, the existing small current will, *when multiplied by the voltage,* indicate that some power is required to drive it. Conversely, the small voltage required to operate a current-operated device will, *when multiplied by the current,* show that power is required to drive it.

A loudspeaker or headset is a current-operated device as can be seen from its rather low impedance. Therefore, in order to obtain maximum performance from these devices when used in tube-type circuits, there must be some way found to transform the rather high output impedance of the tube to the low impedance of the phones or speaker. The simplest method of doing this is to use a transformer. The voltage across the secondary terminals of the transformer is quite low but the cur-

rent available is high—high enough to drive the speaker or low-impedance phones.

Therefore, your question as to how a solid-state device can operate a speaker could have been "how a tube-type device can drive a speaker." Solid-state devices have low output impedances. Therefore, they naturally are fine drivers for loudspeakers and low-impedance headsets. Because of this, there is no need for bulky output transformers to match the impedances of the amplifier to those of the speakers or headsets.

Elimination of "pops" in a music system

Q. When my music system is turned on and the set is at normal listening level, a change of setting of the function switch produces a loud "pop" in the speakers. In addition, when the tape monitor switch is moved, "pops" occur.

How can the "pops" be eliminated? W. C. Harmack, Toledo, Ohio.

A. The problem of the "pops" in your equipment is not a new or a strange one. The source of this problem is that as a circuit is switched in, the coupling capacitor at the output of a piece of equipment connected to the selector switch charges up by way of the grid resistor of the particular stage fed by the arm of the selector switch. The capacitor will charge until the switch is in the proper position—corresponding to a particular amplifier function, such as tuner. This is so because the capacitor has no charging path to ground. The rapid charging of the capacitor looks to the rest of the amplifier just like any other audio signal coming from one or another of the input sources.

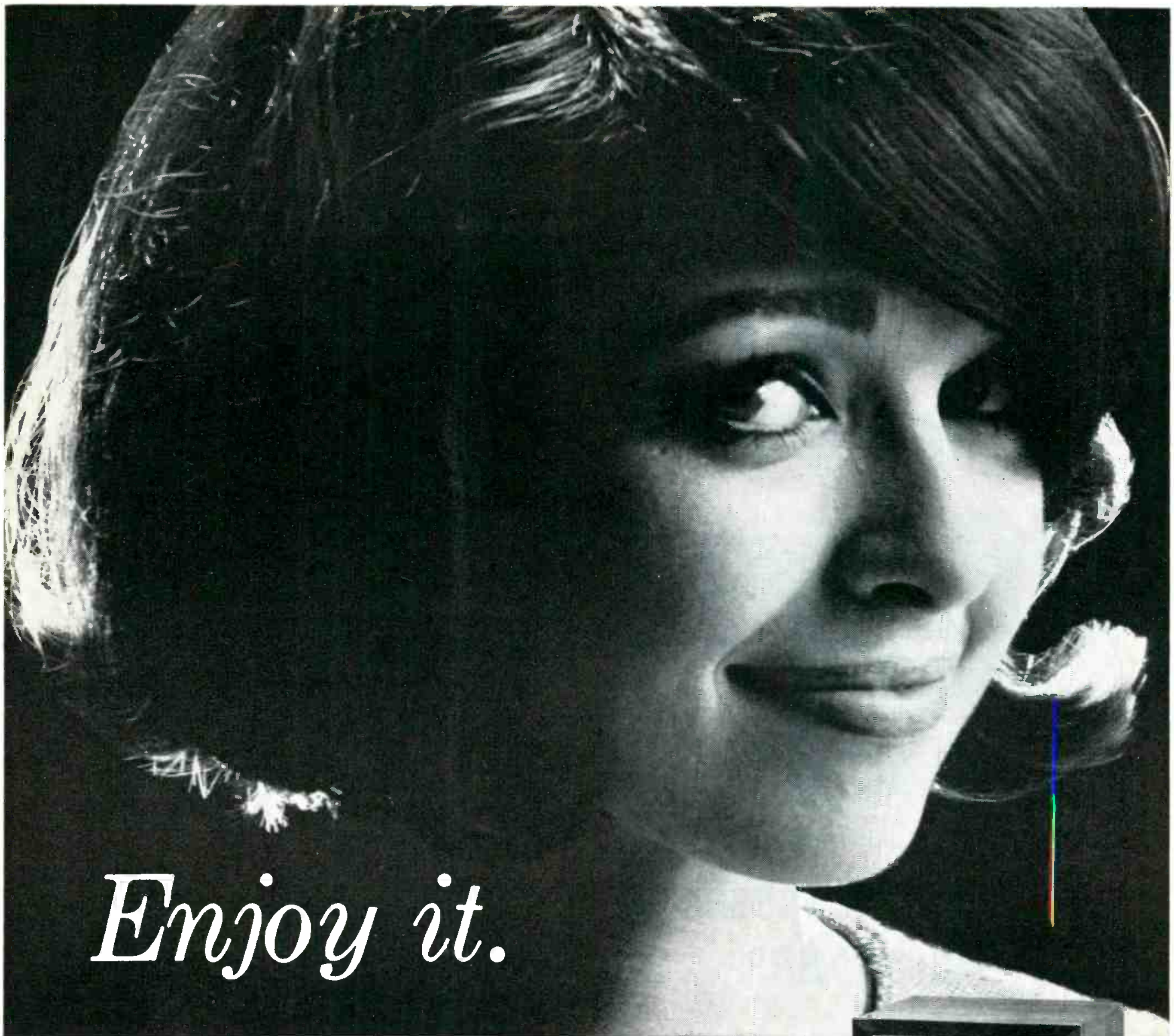
The easiest way to cure your problem is to connect a 5-megohm resistor to each contact representing an input source on the function switch—tuner, tape recorder, phonograph, and so on. Do not connect the arm of this switch to ground. This connection would accomplish nothing. Simply connect each of the contacts to ground via these 5 megohm resistors, one for each contact. Repeat this same process for the monitor switch. If the equipment is stereophonic, then whatever is done on one channel, should be duplicated on the other. In all likelihood your problem will be completely eliminated.

Two Preampliers from a Single Cartridge

Q. Will you please show me how I can split the output from a 5-millivolt monophonic cartridge having an impedance of 200 ohms, and feed it to two preamps, without significant signal loss or impedance mismatch? L. B. Boger, Salisbury, North Carolina.

A. To connect your cartridge to two separate preamps, use a Y-connector. Connect the input of the Y connector to the leads from the tonearm. Connect the two outputs of the Y connector to the appropriate preamp inputs. You can make these connections with no ill effects because the impedance of the cartridge is very low as compared to the impedance of the preamp. Further, the low impedance of the cartridge nullifies any losses in the interconnecting cables.

It may be necessary to run a short, heavy ground strap between the two preamplifier chassis so that hum can be kept to a minimum. Æ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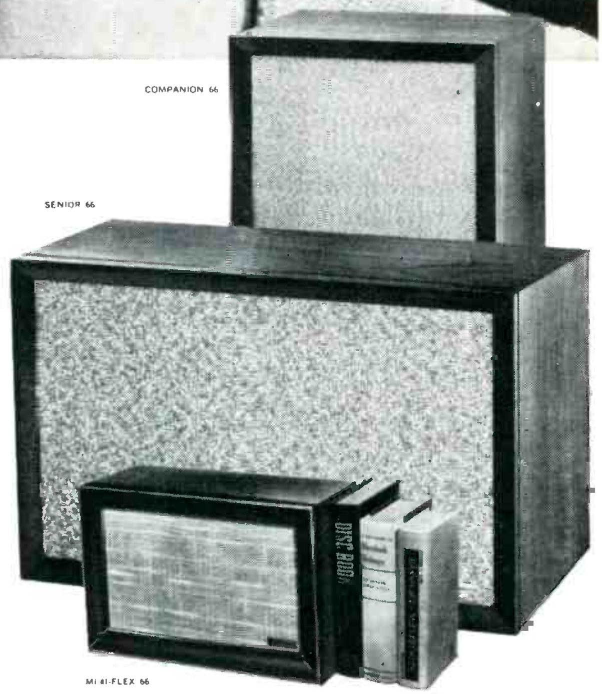
COMPANION 66
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LETTERS

We Get Around

SIR:

I must say that I have been favorably impressed by the reader reaction to my article, "Effect of power-line variation on low-frequency operation of amplifiers," which appeared in the February issue. The replies to the article were for the most part admissions of similar problems and requests for some form of cure.

However, the most remarkable reply to date is the letter received recently. I am told that the *Reader's Digest* gets around a bit—well, so does AUDIO. Keep up the good work. We may have found another common bond between man—Hi Fi.

GEORGE S. LEHSTEN, Chf Engr.,
Alpine Geophysical Associates, Inc.,
Oak St., Norwood, N.J.

The Letter—

SIR:

I read your article in the February issue of AUDIO with great interest, since this is exactly the difficulty I am experiencing at present, with the mains nominally known to be 110v, 50 Hz, but temporarily dropping down to about 85 volts during the evening hours, just when I have time for listening to my records.

I have considered ferro-resonant a.c. regulation and/or stabilization as a means of remedying this situation, but with the lack of data at my disposal there is some uncertainty about this on my part, which is why I am writing you about it. . . .

B. THE,
c/o Djl. Lemboh 78,
Makassar, Indonesia.

(AUDIO has subscribers in many places unfamiliar to most of us, such as Aden, Saudi Arabia, Southern Rhodesia, Nigeria, Liberia, Cyprus, Congo, Thailand, Turkey, Pakistan, Macao, Iran, Israel, and Sarawak, to name a few. ED.)

AUDIO and Others

SIR:

I have enjoyed AUDIO since about 1956. I find the articles interesting, although I have not tried any of the construction projects, but I am looking very seriously at the transistor oscillator which appeared last July. (*We, too, with our usual modifications.* ED.) I am particularly interested in transducer articles since this is still a region for major improvement in technology, perhaps less in speaker design now than in phonograph pickups.

I'd also like to see some basic electronics on transistor operation and use so I can get the "feel" of transistors that I have with vacuum tubes.

Finally, a word about another publication which I don't think is a competitor but apparently you do—*Stereophile*. I should think that Mr. Holt would be allowed to advertise in AUDIO. (*He would, if he wanted to.* ED.) Mr. Holt sometimes generates more heat than light (Something which AUDIO does not do), but there is a need for a publication of that sort in the industry. When I want information on "how to do" or "how does it work," or "try this project at home," I turn to AUDIO, but if I want some controversial reading, *Stereophile* certainly fills the bill, and I think it could do with wider circulation.

GRAHAM B. FULTON
Manchester, Conn.

Back Issues Available

SIR:

I have 75 issues of AUDIO ranging from July, 1959, to October, 1965, all in good condition, which I wish to sell for the special lot price of $16.00 on a first-come, first-served basis. Upon receipt of the $16.00, I will ship via REA Express, transportation charges collect.

If you know of anyone interested in such a compilation of historical and technical know-how, please pass this offer along to him.

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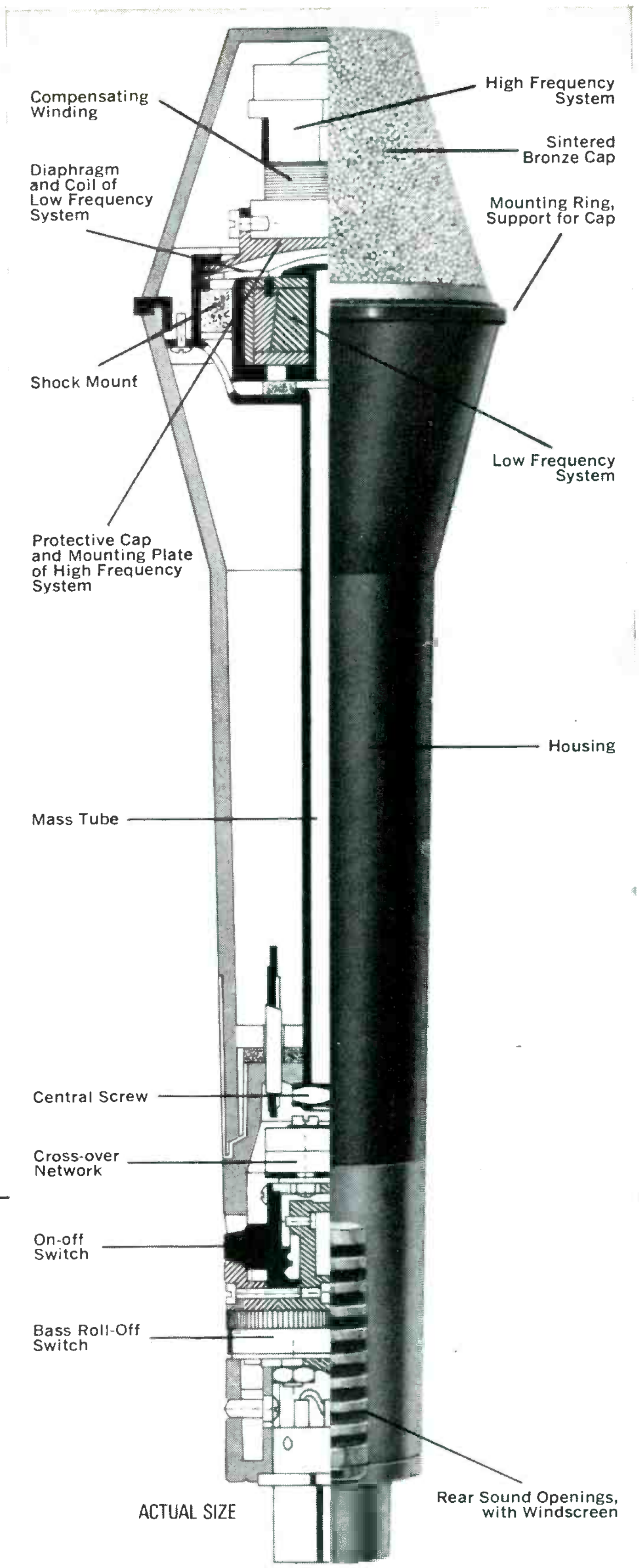
The D-202ES is ruggedly constructed and includes a new sintered bronze cap which functions as a wind and dust screen. Its specifications follow. Its performance is unexcelled. Need we say more? Send today for more information.



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SPECIFICATIONS

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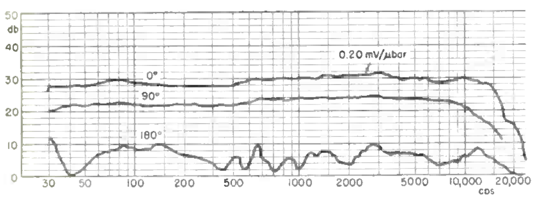

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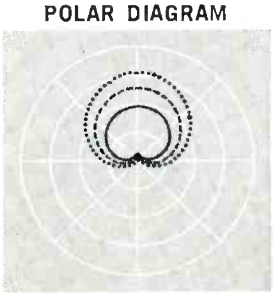

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LIGHT LISTENING

Chester Santon

On a Clear Day You Can See Forever (Original Broadway Cast)

RCA Victor LSOD 2006

The 1965-'66 musicals on Broadway will have to do better than this show if they hope to make any real dent in theatrical history. "On a Clear Day" was expected to kick off the new season in hit form because it is a product of two great names in the Main-Stem firmament. Alan Jay Lerner has provided the book and lyrics. Now that Frederick Loewe, his former partner, is no longer active as a composer, Lerner has teamed with another great Broadway tunesmith for this show. "On a Clear Day" boasts music by Burton Lane, still very well remembered for the great tunes he contributed to "Finian's Rainbow." Although these two had collaborated once before (a 1951 motion picture called "Royal Wedding"), this was to be their first Broadway production. It would be a pleasant duty to greet their latest effort with the cheers their reputation deserves. Instead, "On a Clear Day" spins a tale of extrasensory perception that baffles most of the senses this reviewer can bring to bear in following it. Without the visual benefits of costume change, the home listener is hard put to figure out the action as it shifts between the twentieth and eighteenth centuries. Perhaps the major weakness is a score that can boast only one passable candidate for musical honors, a nice ballad called "Melinda." Too many of the other tunes are trite beyond belief for a composer of Lane's stature. It would seem that my reaction to the show album is shared by some potential ticket buyers in New York City. Despite a healthy advance sale, reduced-price seats for the show became available through local channels the first week in November.

Art Buchwald

Capitol T 2205

Here is one debut on records that should please anyone susceptible to adult humor. Art Buchwald has long been one of the funniest men in daily print. His vocal delivery is equally devastating in this Capitol taping of one of his regular lecture appearances. The college audience at the taping session is with him all the way. Buchwald, a far-from-innocent native of Mt. Vernon, N.Y., first gained fame as a free-wheeling columnist writing from Paris for the *New York Herald Tribune*. His fourteen-year stay in Europe equipped him with an oversized air of detachment which he then proceeded to apply to American foibles upon his return here. By now he's become one of the best political humorists since Will Rogers. Somewhere along the way, Buchwald developed an economy of expression that is the hallmark of truly biting satire. There's hardly a non-working word to be found in the narrative of his career that opens the record or the rest of the topics roasted in the course of the disc. Buchwald's successful debut as a record comedian is helped by a low-pressure style that avoids gags as such while rambling through his slightly outlandish experiences with a disarming air of surprise.

12 Forest Ave., Hastings-on-Hudson, N.Y. 10706

Skitch . . . Tonight!

Columbia CS 9167

Here's a quirk you don't encounter very often in today's dog-eat-dog entertainment world. In this fine recording the Columbia Broadcasting System, through its Columbia Records subsidiary, is boosting the Skitch Henderson orchestra which appears nightly on the "Tonight" TV show seen and heard over the facilities of the National Broadcasting Co. On the basis of this disc, I would say that NBC's arch rival is doing more for the Henderson band with this release than its home network manages in weeks of broadcasting. It is no fault of the band that it finds itself pretty much in the background while Johnny Carson's "Tonight" guests hold forth on the show. There are top men in the band and this record shows it. The smart arrangements prove this orchestra can hold its own with the best of them. Imaginative novelties occupy much of Henderson's attention in this program. His sense of humor flows easiest in "Tootie Flootie," an arrangement for four regular and an alto flute. Equally impressive is the fact that the band doesn't lose any of its sophistication in the "weightier" swing tunes.

Percy Faith Strings: Broadway Bouquet

Columbia Tape CQ 754

It is an established fact among tape (and record) fans that Percy Faith has released some of the finest performances available in the lighter fields of music. The resourcefulness of his arrangements and the manner in which they are executed place him in a unique niche. This happy state of affairs goes back to early days of stereo tape when two-track releases were considered a semi-luxury medium. In recent years Faith's style, along with that of Kostelanetz and a few other Columbia stars, gradually underwent a significant change—a change calculated to make him more popular with the teenage crowd. For a post-teenage reviewer such as this one, it's a pleasure to find him back in his regular style of music making in this wide-ranging collection of Broadway favorites. The sorcery in the handling of high, middle, and low strings tends to

(Continued on page 56)

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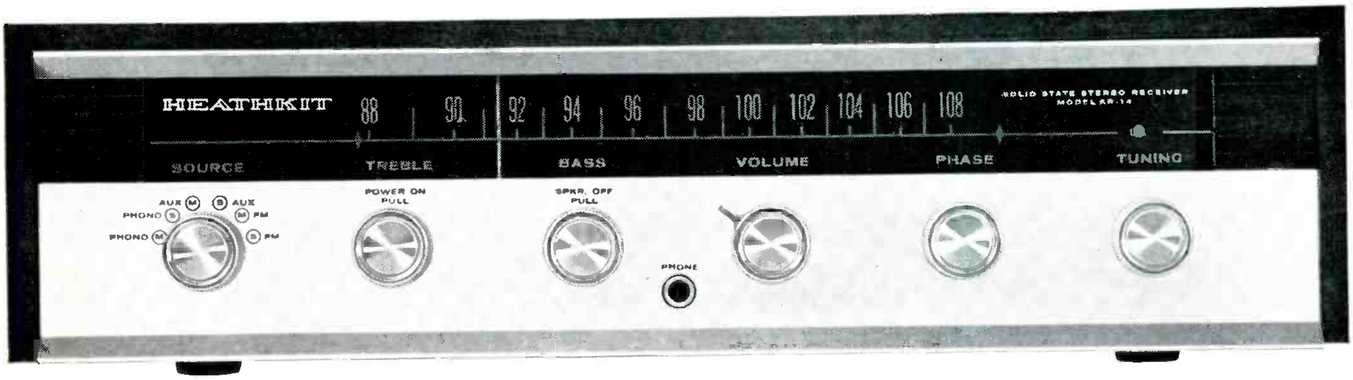

This Remarkable New HEATHKIT® FM Stereo Receiver Has 5 uv Sensitivity, 30 Watts Music Power, All-Transistor Circuitry, and Costs Just $99.95*

Only Your Pocketbook Can Tell The Difference . . . between the superb solid-state performance of this new Heathkit AR-14 Receiver and its low price. Here for the first time is a truly fine stereo receiver for under $100. With *no* component skimping. And *no* circuit short-cuts.

How Can We Do It? First, famous Heath know-how that's been responsible for the best value in hi-fi components since 1949.

Second, you buy direct from the factory (delivered right to your door) . . . *no* middleman profits to add to the price.

Third, you build it yourself in around 20 hours. Thus, labor costs are eliminated. And building it is half the fun. Takes no special skills or knowledge. Simple Heathkit step-by-step instructions and large pictorials show you what to do, and how to do it . . . even how to solder! The tuner "front-end" is already preassembled & aligned. You just assemble two circuit boards, mount some parts on the chassis, and wire the controls. So easy anyone can do it. And what better way to see what it's made of?

Advanced Solid-State Circuitry. The AR-14 uses 31 transistors and 11 diodes for that clean "transistor sound", cool operation and long life . . . *no* tube deterioration.

Its new transformerless output circuit is known as complementary design. Each channel uses two output transistors—a PNP type, and an NPN type—complementary. This has proven superior, exhibiting lower phase shift, wider response, and lower distortion. All power transistors are adequately heat-sinked for long life.

The FM multiplex tuner circuit uses a 3-transistor "front-end" and a 4-stage IF section.

15-50,000 cps Response . . . Sensitive Tuning. Two preamplifiers and two power amps provide a frequency-power response of ± 1 db from 15 to 50,000 cps . . . and at *full power* in both channels, thanks to a generous power supply design. And there's plenty of power to drive all but the most inefficient speaker system . . . 30 watts IHF music power or 20 watts RMS at 4 and 8 ohms output, slightly less at 16 ohms. Its sensitive FM tuner eagerly seeks stations, far and near . . . and you'll like the easy way the flywheel dial glides across the band. For beat-free tape recording, the tuner outputs are filtered.

Front-Panel Controls For Full Command! Include a 6-position source switch to handle the tuner, plus your tapes and records, stereo or mono; dual-concentric volume control to individually adjust each channel; a tandem treble control with push-pull AC power switch; tandem bass control with push-pull speaker on-off switch for use when listening with stereo headphones from the convenient front panel jack; stereo phase control for 45 db or better separation; and automatic stereo indicator light.

Bookshelf Size, 3-Way Installation. Measures just 3⅞" H x 15¼" W x 12" D . . . a boon for limited spaces. Its handsome anodized extruded aluminum front panel fits neatly in a wall, your custom cabinet, or in either optional Heath walnut veneer or beige steel cabinets. Marry it with two speakers and enjoy. Use the coupon to order yours now.

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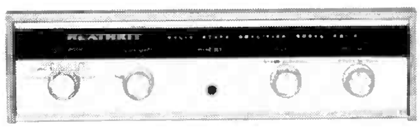

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17 transistors, 6 diodes. Performance and features equal to AR-14's amplifier section. Assembles in around 10 hours.
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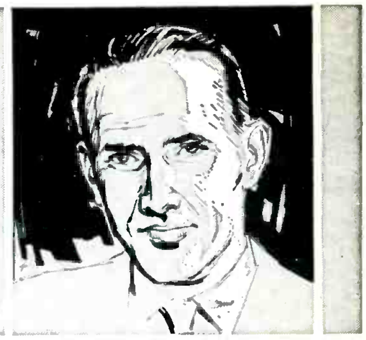
AUDIO ETC.

Edward Tatnall Canby



SOLID-STATE MICRO-VALVE

It's been quite a while since I started using the pickup which is surely the most "revolutionary" in the last quarter-century—at least in principle, the one which applies a transistor valve to the transducing job that starts at the stylus tip. The Euphonics Miniconic phono cartridge is the gadget and, though it was not exactly a brand new sensation when I got to it, I felt immediately that here was a fundamental innovation of a sort I could not afford to pass by. For in this cartridge the stylus operates not as a generator, but as a true valve. This is in effect an amplifier-transducer, the signal input a physical vibration, the outgoing signal a modulated biasing current. The solid-state element—called coyly the Pixie—is thus basically unlike the equally solid-state ceramic or crystal unit, which is a strain-sensitive *generator*, not a strain-sensitive *valve*.

Now this is a wow of an idea! And (as Euphonics says, with no more than the usual commercial modesty), as modern as tomorrow. It brings a variant of the transistor, a silicon element, to still one more basic segment of the hi-fi system. In truth, it does provide some mighty impressive advantages. Too many to go into in detail (and, as always, this is done much better by my more technical colleagues)—but what impressed me especially was (a) the extremely small size of the silicon unit itself, the business end of which is actually smaller than the diamond stylus, and (b) in contrast, the remarkable over all potency of this bias-current arrangement, which produces a signal out of the stylus motion with (they say) some 10,000 times the efficiency of a magnetic. Extremely tiny physical size, relatively whopping output! That's something. And most impressive of all is the principle, which at last frees the stylus from the onerous job of electrical *generator*. Now, it merely operates an electrical valve—it is the mechanical "grid" which controls "plate" current.

That is the same basic step, you see, as we took 'way back in the Twenties when we removed the mechanical stylus, the old phonograph "needle," from the nasty job of acoustic sound-generation. Until then, acoustic output depended directly on stylus energy. Afterwards, we could amplify a given stylus signal to any degree, from *outside* sources of energy. But, through all these succeeding years, the pickup cartridge itself has remained a basic generator, electrical instead of mechanical.

Now—Euphonics reduces mechanical work to the ultimate minimum, or almost. The tiny physical motion of the unusually small stylus assembly, controlling the silicon valve, is all that is left. ("Beam-of-light" or electron-beam groove tracking? That would eliminate *all* physical motion from the input end of the disc system; we often hear of it, but nobody yet seems to have figured how to track a groove without physical contact. Maybe somebody will invent a groove "carriage," which will ride along the groove and carry on itself an electron beam or something, aimed into the groove which is being tracked but—somehow—uncoupled from its physical vibrations. Ugh. Makes me see double just to think about it.)

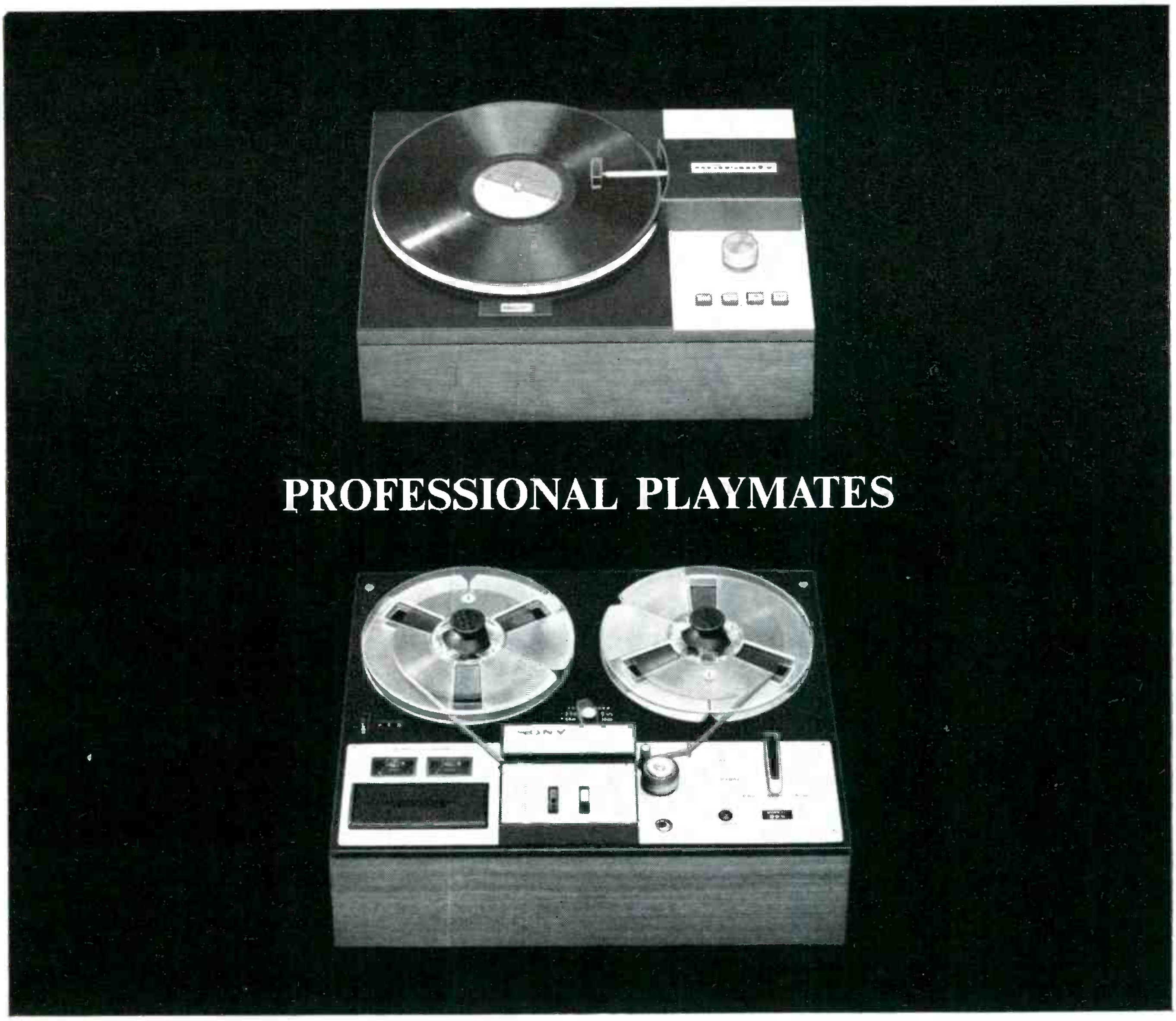
So—this would seem to be "it," this Euphonics principle. More power to the idea.

It remains, when such an idea is put into practice, clothe it in reality. This the Euphonics people have done, in the two basic senses. First—the unit itself, the silicon valve and the stylus attached to it, mounted in a workable cartridge head. Plus optional power supply (until the bias or "plate" supply is available on standard amplifiers). Second, the associated arm and assorted optional equipment to adapt the arrangement to existing home or other playing systems.

Sometimes I think these relatively unexciting but necessary after-jobs are the most important of all—for they make or break the commercial embodiment of an idea. And the more radically new the idea itself, the tougher is the assignment.

I won't soon forget, for instance, the early and pioneering Vico transistor amplifiers, in which the new revolution in solid-state amplification was embodied in such grotesquely miscalculated inputs, outputs, impedances, and what-not that it was very nearly impossible to get a usable signal from a standard pickup to a standard loudspeaker. Just minor petty matters, for the most part, and a lack of experience in the crude realities of the hi-fi field as it then was.

All I need to say, then, is that—all things considered—the Euphonic people have done a very good job in this vital respect. I received a fairly early sample of their complete integrated arm-and-cartridge, the Miniconic TK-15-LS (the fancier of two alternative cartridge types—the P models have conical instead of bi-radial, or elliptical, tip) and I experienced merely the normal Canby quantity of agony in the throes of installation and record listening.



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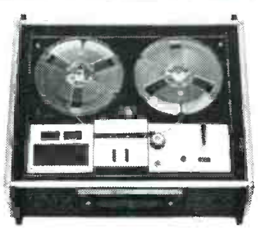

Portable Model 350C

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For example, the instant my big fingers touched the removable plug-in cartridge head, it came out of the arm socket—but in the wrong place. The plugged part, four neat little flat contacts, stayed in the shell. The rest pulled out—and the tiny wires . . . well, luckily I had my most skillful assistant on hand and he was able to micro-solder things back together again. Just one of those little details. Then I discovered an odd faculty of the lightweight tubular arm, which is mounted via a single short protruding bearing rod that sits in a hole, with a thinner side rod which acts as a detent to keep the arm from turning too far in either direction (yes, it is adjustable). The front end, when not working, goes into an arm rest of U-shaped plastic and is clamped there by light friction. Unfortunately for me, every time I tried to push the arm into this arm-rest socket the rear end heaved itself up and out of its bearing hole and the whole darned arm flopped over, dead, with the tiny connecting wires dangling dangerously loose! Again, a minor matter (though a miscalculation on somebody's part) and I soon learned to live with this type of accident-proneness.

But what does it sound like? Well, I hardly need say that both my ears and that of corroborating witnesses testify as to the expected high quality sound available from this silicon-valve system. Clean—but yes. It is plus-or-minus 2 db 20-to-20,000 in the specs, and responds from "d.c. to beyond 50 kHz." It definitely sounds that way.

Moreover, the entire stylus unit is extremely light in mass—the diamond is held direct (nude is the lovely word) rather than mounted in a bulky metal surround. That most definitely is a fine idea, and precisely what every cartridge-maker works towards, as well as he is able to with his particular design. The non-generator stylus, though, can be smaller than most—and thus sound cleaner. And the system itself is simpler than most, as well as lighter. Even simpler than a ceramic generator array.

In actual practice I found that the main working problems with this pickup assembly were like those of every other very modern lightweight, ultra-wide-range unit. The thing sticks to your fingers when you try to put it on the record, it's so light. It fouls itself in microscopic swirls of dust and skips grooves when it hits the tiniest trace of a grit-mountain—just like most new pickups today. And I did notice that with an output down to d.c., this cartridge is likely to show up any and all low-frequency deficiencies of a mechanical sort. Acoustical feedback through the floor, for instance, gives a beautiful and destructive b-b-b-blat at around 8 Hz if I accidentally turn the volume 'way up, through my 100-100 solid-state amplifier. Phew!

These are not in any way to be considered serious impediments. And the price for the whole system is decidedly reasonable—this is *not* an inherently expensive type of pickup, except insofar as it is polished up to laboratory tolerances of

(*Continued on page 57*)

ABOUT MUSIC

Harold Lawrence

KABI LARETEI—A TALKING PIANIST

The woman in the red dress who walked onto the stage of Carnegie Hall had the sharply defined features of a Katharine Hepburn and the erect posture of a ballet dancer. Turning to the audience to acknowledge the applause, she announced, in a conversational tone that seemed to reduce the large auditorium to an intimate salon, that the work she was to perform—Hindemith's *Ludus Tonalis*—had been described as a modern counterpart of Bach's *Well-Tempered Clavier*. Familiar to advanced piano students everywhere, it nevertheless remained unknown to the average music lover. She was determined to remedy the situation. Later she would invite the audience to ask her questions about the music and would repeat sections of the work upon request.

More applause. Then Käbi Laretei, wife of the celebrated film and theatrical director Ingmar Bergman, sat down and played through the hour-long collection of fugues and interludes with no more than a two-minute break. According to the *New York Times*, her performance revealed "the varieties of mood and appeal offered by the fugues [and made much] of the virtuosity, wit, winsomeness, and tenderness contained in the interludes." The audience responded warmly. After several curtain calls, Miss Laretei held up her hand to stop the applause. In a light clear voice, she said that, according to the printed program, the question-and-answer period was to follow the intermission; but would the audience like to dispense with the break and go on with the program. "No intermission!" came the shouted reply. Beaming, the pianist invited those in the rear of the orchestra to move forward and occupy seats closer to the stage. (The hall was only partially full due to the recently concluded New York newspaper strike.)

*152 E. 62nd St., New York, N.Y. 10021

It should be pointed out at once that such a procedure is highly unorthodox at Carnegie Hall where artists seldom speak to the audience except to announce encores in telegraphic style: "Chopin . . . Mazurka . . . A minor." Miss Laretei's break with recital-hall tradition made for a refreshing musical evening.

Good looks, poise, and solid pianistic talent have made Käbi Laretei the most popular musical performer on Swedish television . . . a sort of female Leonard Bernstein. She enjoys sharing her enthusiasms with her audience, demonstrating passages to point up her talks, and generally advancing the cause of classical music. But she is more than an articulate exponent of her art. A pupil of the late Edwin Fischer, Miss Laretei has impressive credentials as a concert artist. She made her debut in Stockholm in 1946, and soon began touring with principal orchestras in Sweden and throughout Scandinavia. Engagements as a recitalist and with leading orchestras in Europe and England consolidated her position on the musical scene. But it was during an English tour that Miss Laretei first discovered her special abilities as a "talking pianist." Her television debut on the BBC was a resounding success and led to a series of TV programs in Stockholm. Since then, Miss Laretei has become known in Sweden as "Mrs. Music" because of her programs of talk and music for children and adults, and her frequent recitals and concert appearances.

Despite her broad concert and TV experience, Miss Laretei has made few recordings. But when Philips Records engaged her to record *Ludus Tonalis* two days after her Carnegie Hall performance, no one expected her to suffer from mike fright. Early on the afternoon of the first session, she arrived at the studio for a warm-up. Miss Laretei wanted to accustom herself to the acoustics of the studio, formerly the ballroom of the Great Northern Hotel. Apart from the construction of a glass-paneled control booth, and coats of white paint, the hall had not been "treated"; the sconces, chandeliers, fluted columns, and hardwood floor combined to give the room a bright, reverberant sound, more "live" than that of most studios.

"Too much glitter, not enough warmth," was Miss Laretei's reaction after playing a movement from the work we were to record. Engineers and technicians rolled out bright crimson carpets at various strategic places on the polished floor. After another test, Miss Laretei said that the damping had improved the sound considerably, but that it still seemed blurred from her vantage point. A rug was moved closer to the piano, and another placed under the piano stool.

Inside the control booth, two three-track Ampex tape machines were loaded and ready to go. The three Telefunken microphones (♯ 201) were arrayed evenly along the side of the piano and between six and seven feet above the floor. Miss Laretei was asked to play the Praeludium, which covers a wide dynamic range from the brilliant opening to the toccata-like succession of broken chords, the lyrical Arioso, and the solemn, broad conclusion.

The first thing a recording producer listens for as the sound emerges from the monitor speakers is balance. Are all the registers of the piano in proper perspective? In the first page of the music, it was apparent that the sound was bass-shy. Engineer Robert Eberenz lowered the right microphone, moved the boom forward slightly, and the bass notes became noticeably rounder and had more body. We turned to the treble: not enough presence; more reflected than direct sound. The left microphone was brought closer to the instrument. Now the mechanical action of the pedals and hammers was obtrusive and the pickup brittle. A half-way point was selected (a matter of inches) and instantly the sound improved: we had gained presence without the ear-on-the-strings percussiveness that mars so many piano recordings The same procedure was repeated for the middle register, after which we listened to the over-all balance. The tests consumed 45 minutes of our

(*Continued on page 67*)

The author discusses the score with Miss Laretei during the recording session.

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EDITOR'S REVIEW

Whenever we think of it in time to make the January issue, we like to list all of the high fidelity shows which are definitely scheduled for the coming year by the time the issue goes to press. In order to make such a listing complete, we must include those shows all over the world, as well as the more familiar ones in the U.S. In addition to those shows which are strictly hi-fi affairs, there are a few others which are of general interest to the electronics world, and those, too, we list.

The first hi-fi show scheduled is in Philadelphia, at the Benjamin Franklin Hotel, on February 18, 19, and 20. We'll be there.

Next comes the Festival du Son, the fifth hi fi show to be held in Paris, from March 10 to 15. This is followed by the Los Angeles High Fidelity Music Show, which takes place from March 27 through April 3. Jumping over the Atlantic again, we encounter the International Audio Fair and High Fidelity Show at the Russell Hotel in London, April 14 to 17. Coincidental with this is the U.S. Department of Commerce exhibit of U.S. products, to be held at the U.S. Trade Center in London from April 14 to 22. Some 37 brands of American-made hi-fi gear are slated to be shown to the British for the first time by a U.S. Exhibit.

Audio will most certainly be at the U.S. Show in London, since we will include, in our March issue, the complete program of the exhibit, together with some historical material about the U.S. shows held so far in Frankfort, Milan, Tokyo, and Bangkok. And since we will be on the scene, we will certainly be able to visit the local event as well. With two shows vying for the attention of the British buff, they should go all out to be as interesting and attractive as possible.

Back to San Francisco on April 18 for the IHF show which lasts from the 18th to the 25th. Obviously, Audio will have to be represented there, as well as in Los Angeles.

Next in the list of hi-fi shows is the BIG one—New York. This is scheduled for the fall, and is the most looked-forward-to show of the year. This one is scheduled, but where and when are not yet specified.

In addition to the hi-fi shows, but of interest to electronic engineers, there is the Salon International des Composants Electroniques, to be held in Paris from Feb. 3 to 8. And the enormous Messe in Hanover takes place from April 30 to May 8. And of course we mustn't overlook the IEEE show in New York from March 21 to 24. National Electronics Week, the successor to the old Chicago "Parts Show" is scheduled for May 30 to June 5 in San Francisco. Then comes the Music Show—increasingly important to the audio industry—to be held in Chicago from July 10 to July 14.

That is an impressive list—it would appear that the coming year will signal the coming of age of the entire high fidelity business. We look forward to an interesting 12 months, which should bring a greater awareness of high fidelity and an enormous upsurge of video home recording.

GERMAN TRIP

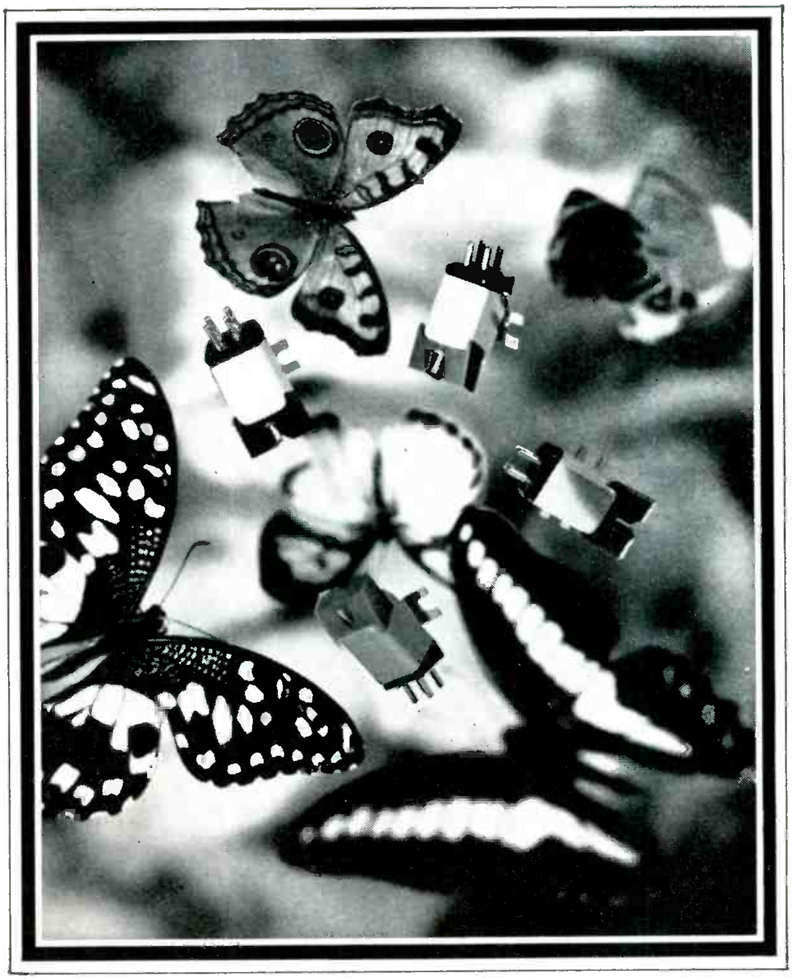
There is a possibility—and so far, only that—that a trip will be offered through Germany to visit a number of audio equipment manufacturers, much in the footsteps of our own visit there in 1964. This will be made available to anybody interested (if it does materialize) and it is expected that it will follow immediately the U.S. Department of Commerce exhibit of U.S. hi-fi gear in London. As soon as any further information is available, you may be sure that you will read about it here.

HOME VIDEO RECORDING

Although video recorders are "just around the corner," even though some are now available, we find an enormous interest in the whole subject. Primarily the recorder mechanism is the same breed of device as an audio recorder, with the added feature of the rotating head. As a matter of fact, the first one we got our hands on got rather a beating in shipment, but we had it running right off the bat without ever looking into the instruction book—a practice we don't recommend. That is to say, we had it running and recording audio, and it was obviously playing back, but no picture. The record circuitry had been damaged. The monitor worked, and we could tell that the playback was operating. We are trying to get over the point of its handling essentially like any other recorder.

Basically the techniques do not differ appreciably from those of an audio recorder, as far as the operator needs to know. But the whole operation opens up a new vista of things to do—anyone who has photography as a second hobby will be fascinated by what a video recorder can do, and how well it does it. We haven't begun to exhaust the possibilities—one of our first interests now is the construction of a video camera—just for the fun of it.

And that is, after all, why we have hobbies—something we can "work" with (read that "play," if you like) and derive enjoyment from it. We are certainly enthusiastic over the prospects of home video, and we think you will be, too.



Compare these Sherwood features and specs! ALL-*SILICON* reliability. Noise-threshold-gated automatic FM Stereo/mono switching, FM stereo light, zero-center tuning meter, FM interchannel hush adjustment, front-panel stereo headphone jack, rocker-action switches for tape monitor, mono/stereo, noise filter, speaker disconnect and loudness contour. 100 watts music power (8 ohms) @ 0.3% harm distortion. IM distortion 0.1% @ 10 watts or less. Power bandwidth 12-35,000 cps. Phono sens. 1.8 mv. Hum and noise (phono) —70 db. FM sens. (IHF) 1.6 μv for 30 db quieting. FM signal-to-noise: 70 db. Capture ratio: 2.4 db. Drift ±.01%. 40 silicon transistors plus 14 silicon diodes and rectifiers. Size: 16½ x 4½ x 14 in. dp.

Model	V-Vacuum Tube S-ALL-SILICON T-Germanium Transistor	Power (IHF) 2 channels 8 ohms Watts	Max. IM Distortion Below 10 watts	FM Sensitivity Microvolts	Price	Dollars/Watt
Sherwood S-8800	S	100	0.10%	1.6	$ 359.50	$ 3.60
Altec 711	S	100	0.15%	2.2	378.00	3.78
Bogen RT 8000	T	70 (4Ω)	0.3%	2.5	319.95	4.57
Dyna FM-3, PAS-3, & S-70	V	90	0.1%	4.0	394.85	4.38
Fisher 600 T	V & T	120	1.6%*	1.8*	459.50	3.82
Harman-Kardon SR-900	T	75 (4Ω)	0.9%*	3.3*	429.00	5.61
McIntosh MR71 & MA230	V & T	88	0.25%*	1.8*	748.00	8.50
Marantz 8B, 7, & 10B	V	75*	0.2%*	2.0	1170.00	15.60
Scott 348	V & T	100	0.5%	1.9	479.95	4.79

Reference "T" (above) may include some silicon transistors.
Figures above are manufacturers' published specifications except (*) which are published test findings.

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Push-Pull Drive from a Split-Load Stage.

GEORGE FLETCHER COOPER

To obtain optimum performance in transistor driver stages, it is necessary to make thorough analytical checks on the various operating conditions, investigating drive current and voltage, and the resulting source impedances.

THE USE OF A LOW-IMPEDANCE DRIVER to provide a wide frequency response from output transistors of limited beta cut-off frequency is reported to give an improved frequency response. A very elementary analysis shows that this is true, but as I pointed out in an article recently (Transistor Output Stages AUDIO, Oct. 1965), the advantage may be just an illusion. My reasons for suggesting this are that during the interval between the end of the alpha rise-time and the end of the beta rise-time the driver stage will need to produce an excessive amount of drive current. On the other hand the drive current at the high frequencies with which we are concerned will normally be very low and this may save the situation.

If we accept the limitations of this kind of system, which certainly offers the possibility of producing power output stages with the cheapest power transistors, should we limit ourselves to the use of the common-collector driver? Once we accept the fact that we may well need a rather large drive current we see that the common-collector stage in the form to which a correspondent has drawn my attention is not particularly convenient. This form uses a 1:1:1 transformer to drive the push-pull output stage and is almost equivalent to using a p-n-p/n-p-n phase-splitter as a common-collector drive pair except that the single transistor must be operated in class A.

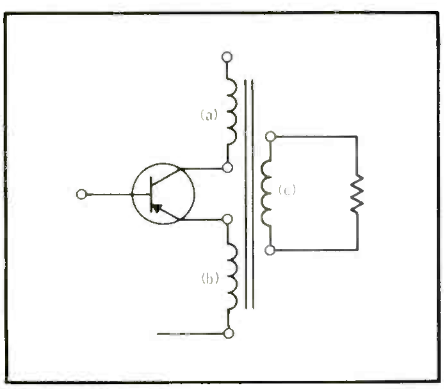

Fig. 1. Basic form of split-load stage.

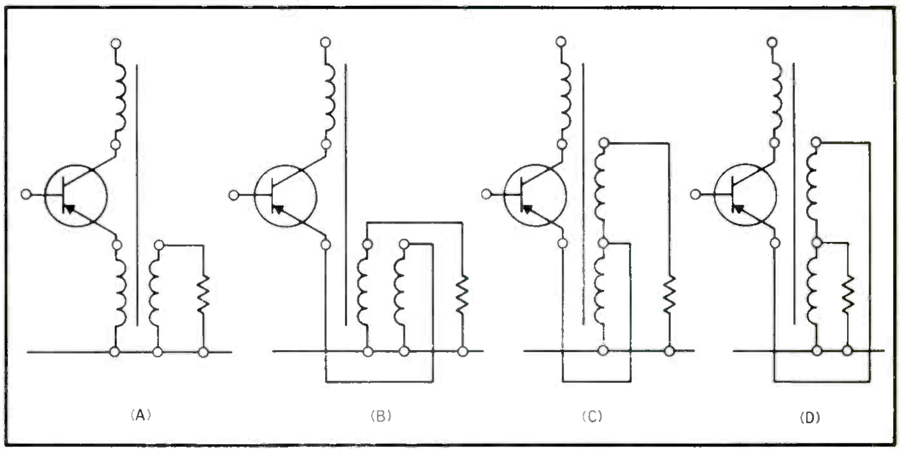

Fig. 2. The split-load stage displayed as a feedback amplifier.

The more I think about this circuit the less I like it. We have gained the very tight coupling of the trifilar transformer at the cost of running the drive transistor inefficiently with a particularly non-optimum load. One day I must take up a remark by Mr. Crowhurst in the May, 1962, issue of AUDIO on this matter. The point at issue, however, is that we need to get plenty of power from our driver and we must try to give it a reasonable load. As usual, a grounded-emitter stage will give us an excessive source impedance while a grounded-collector stage needs too much voltage drive to operate efficiently.

This situation is one which calls for the use of a split-load circuit. There is nothing which is fundamentally special about this circuit, nothing which justifies giving it an individual name, except the fact that in applying feedback around a conventional stage the circuit has been so simplified that it is not practicable to break the system down into an amplifier and a feedback path which can be treated separately. Since we must analyze the circuit as a simple unit and since we must write something at the head of the page, this circuit has a name.

General Circuit Configuration

The basic circuit is drawn in *Fig.* 1. The load is coupled to transformer windings in both the emitter and collector circuits of the transistor and is thus split, as it were, between the emitter and collector. The transistor is therefore acting as something between a common-emitter stage, which would have all the load at the collector, and a common-collector stage, which would have all the load at the emitter. The properties of the circuit are thus likely to be intermediate between the two conventional forms and we shall see that this is indeed the case. The use of the split-load technique gives us a degree of freedom in design which is of great value. We often face the difficulty that the common-emitter stage will not do what we want, but we cannot provide the drive for a common collector stage.

There does not seem to be very much published about the use of the split-load stage, and what little there is appears to be concerned only with its use in the output of an amplifier. One short note concerns the split-load arrangement in a class-B stage and points out its advantages under conditions of low supply voltage and low temperature in reducing cross-over distortion. The discussion of low battery voltage is based on

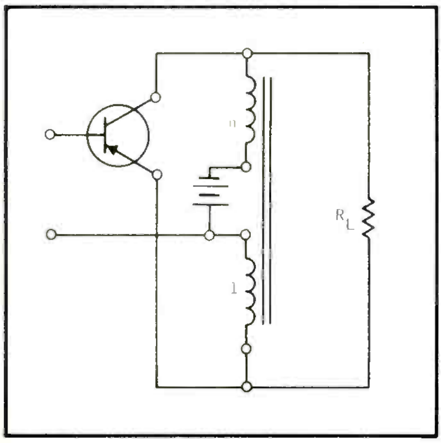

Fig. 3. This way of drawing the circuit keeps the mathematics simple.

the use of a simple fixed voltage-divided chain for the bias. This system will obviously give too little bias if the supply battery is low and we shall get the familiar low-gain cross-over distortion which, because the gain is low, is not greatly improved by the use of negative feedback.

Since the split-load stage demands a larger voltage swing at the bases of the transistors, the origin distortion is a smaller part of the whole signal. From the point of view of a designer who is only interested in his paper figures this can be advantageous. We must remember, however, that it was pointed out quite a long time ago, by Licklider, that this particular kind of distortion sounds much worse than the figures suggest. The reason for this is that the *amount* of distortion is constant and thus the percentage of distortion for small signals is very high even though for a full-load signal the distortion figures may seem quite satisfactory. All that we achieve by using the split-load circuit is a lowering of the level at which the distortion is really unpleasant: pianissimo instead of piano becomes badissimo instead of bad.

In order to analyze the split-load circuit we may begin by rearranging it to see if it has any easily recognized characteristics. We can ground one side of the load, as we have done at (A) in *Fig.* 2, and then simply re-draw the emitter winding of the transformer in a different part of the paper, as we have done at (B) in the same Figure. We do not know whether the load winding or the emitter winding has the more turns but in either event we can reduce the right side of the transformer to an auto-transformer of the form shown at (C) and (D). This configuration shows very simply the way in which we get negative feedback of a fraction (which may be improper) of the load voltage applied in series with the input of a common-emitter stage. We know that this will increase the input impedance and reduce the output impedance. We see that we are likely to be rewarded in our study of the circuit, for both these features are valuable.

When we begin the analysis there are several ways in which we can conveniently arrange the symbols on the transformer windings. We can carry through the whole analysis in terms of the turns *a, b, and c,* shown in *Fig.* 1; we can get a slightly simpler outlook if we take the transformer as having the ratios *m, (1–m),* and *n.* The simplest form of all is based on the fact that come what may the load seen by the transistor is $[(a+b)/c]^2 R$. Provided we make allowance for this impedance transformation in any practical case, we can use a straightforward one-to-one transformer with this load impedance *R* for our theory.

Instead of taking the split as being at a point *m* from one end, which would give ratios of *m, 1–m,* and *1,* it seems to me to be preferable to keep the minus sign out of the ratios. I suspect that this belief that it makes the mathematics easier is completely without foundation, but you must believe in something and so long as no one else believes it you have not much chance of starting any trouble. Anyway, this choice of symbols looks quite reasonable if we draw the circuit in the form shown in *Fig.* 3.

Which Equivalent Circuit

The next question is which transistor equivalent circuit we should use. It seems that if we are to make some savage approximation later we can best use the hybrid-π circuit, especially as the approximation leads us towards the sort of data we are likely to have on a transistor which is not fully documented. The over-all circuit then takes the form shown in *Fig.* 4, in which the transistor "generator" is characterized by the transconductance, G_m.

In the study of transistor circuits we can say that every equivalent circuit implies its own approximations or, if you prefer it, every set of approximations from which we build up an equivalent circuit shapes the final equivalent. Let us do some cutting down in *Fig.* 4 and see where it will lead us. To begin with let us take it that this transistor is fed from a low-impedance source. We know that $r_{bb'}$ is usually low and thus the feedback through $r_{b'c}$ cannot have much effect. We can therefore ignore this term completely in our rough calculations.

We can be pretty certain that the input resistance which is now right across the tap of the transformer, will not absorb any substantial amount of power from the output. In my time I've had a few headaches from the need to put power into the feedback circuit, though always in designing amplifiers which were pushed to produce ten milliwatts, but this is not one of the topics you find taken into account in any of the texts. Combining all these simplifications we can reduce the circuit to the form shown in *Fig.* 5.

There is even one further simplification which we can make, for we know that the pentode-like characteristics of transistors will normally let us work with a load resistance far below the value of the internal impedance. When the circuit is used as a driver stage we shall probably be using a load which is less than the optimum and the argument for dropping is reinforced. The results cannot fail to be pretty simple. We shall drive a current

$I = g_m V_{b'e}$ through the load R and thus generate a voltage $V = g_m R_L \cdot V_{b'e}$.

The value of $V_{b'e}$ will be the difference between the generator voltage V_o and the feedback voltage $V/(1+n)$, so that we have

$$V_{b'e} = V_o - V/(1+n)$$

from which

$$V = g_m R_L V_o - g_m R_L V/(1+n)$$

$$V = V_o g_m R_L / [1 + g_m R_L/(1+n)].$$

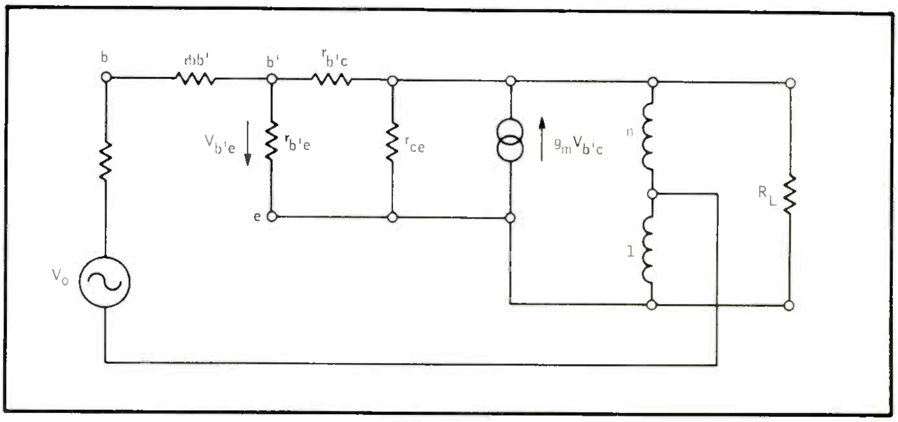

Fig. 4. The full equivalent circuit of the split-load driver stage.

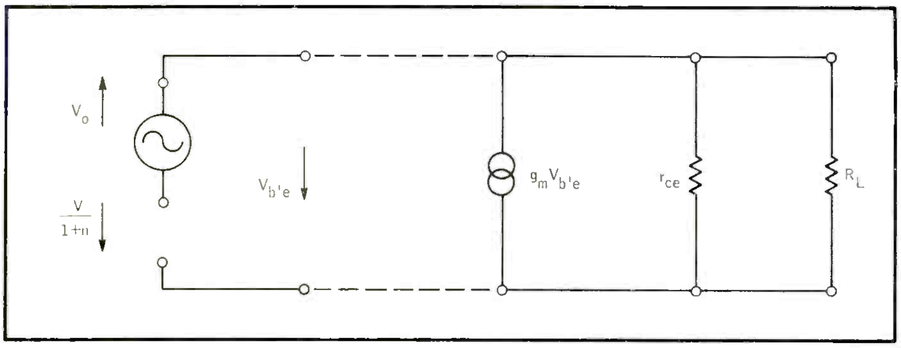

Fig. 5. The approximate equivalent circuit.

Although this equation is of considerable interest it is not what we are concerned with immediately. After all, we are looking for a low-impedance device to act as a driver. Let us force a current I_o into the output terminals. The current which flows through R_L will be made up of I_o and the current from the generator $g_m V_{b'e}$, so that we shall have a total current

$I = I_o + g_m V_{b'e}$

and we shall produce across the load resistance R_L a voltage

$V = IR_L = (Z_o + g_m V_{b'e}) R_L$.

But we must have

$V_{b'e} = -V/(1 + n)$, *since* $V_o = 0$ for this experiment,

$V = [I_o - g_m V/1 + n)] R_L$

or $V [1 + g_m R_L/(1 + n)] = I_o R_L$

from which $V/I_o = R_L/(1 + g_m R_L/(1 + n))$.

This, of course, is the impedance seen at the output terminals. If we make n go to infinity it is just R_L, and if we make $n = 0$ it is just $R_L/(1 + g_m R_L)$ which is the usual form for an emitter follower. We can see this best by taking the output conductance I_o/V, which is $\left(\frac{1}{R_L} + \frac{g_m}{1+n}\right)$. Immediately we see that this is the parallel combination of the load resistance itself, and the actual transistor circuit, which produces a conductance term $g_m/(1 + n)$ corresponding to an impedance of $(1 + n/g_m$. We can use this very simple form until we start to run up towards the value of r_{ce}. For one small transistor, chosen only because it is where the book happens to fall open, we have $I_m = 40$ mA/V and $r_{ce} = 50{,}000$ ohms. With a value of $1/g_m = 25$ ohms we should be using $n = 1000$ or more before we worried about r_{ce}, and that is not at all the sort of value we shall be considering. Even $r_{b'e}$ at 1250 ohms will not be likely to raise any problems, even when n is small, and rather surprisingly this is in fact a more significant term. If we explore the other approximations we find that so long as n is a reasonably small number they are all quite proper simplifications which introduce errors very much smaller than the differences between transistors of the same type.

Practical Example

In order to keep our ideas under control it is probably useful to pick up some fairly real numbers. One moderately small transistor, with a rating just below half-a-watt at 45° C and a current limit of 1A I find has a transconductance of rather more than 1 A/volt and if we take $1/g_m$ as 1 ohm we shall be on the safe side because most transistors of this type will give us lower impedances. This type of transistor is probably capable of driving something like a 10-watt push-pull stage and for this we should need a drive at each base of some 50 mA and perhaps 400-500 mV. We might even go higher with the voltage drive because we might choose to put some resistance in the emitter leads of our power transistors.

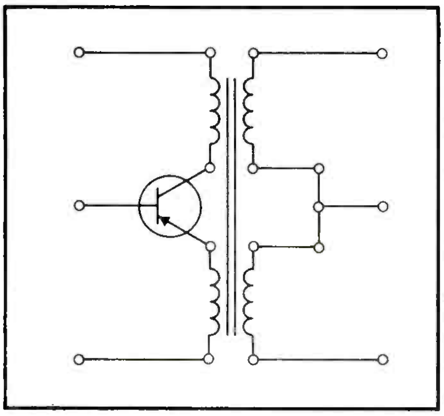

Fig. 6. Quadrifilar-wound push-pull driver.

I am not designing an amplifier here, of course, but looking at the sort of amplifier which might come out from the use of particular transistors. What I have to consider is the best way of using a driver with a 1-ohm impedance to feed a power transistor with a 10-ohm input impedance. This 10 ohms is the standard grounded-emitter input impedance and we might consider that the 1-ohm source impedance was relatively small in comparison. However, we saw in the previous article that what we mean by small in this connection is small compared with the input impedance in the grounded-base connection. Even if beta were as low as 20 this would give us an input impedance of only one-half ohm.

Even a power transistor with a transconductance of 10 A/V connected as a pure emitter follower will not really provide an impedance which from the point of view of distortion can be called low in this context. We may, however, digress to notice that if we choose a value of $n = -1$ in the theoretical expression above we shall apparently get a source impedance of zero. It seems when we go on to apply this to our practical circuit that this is very much a nonsense, because the autotransformer in *Fig.* 4 becomes a short-circuit of its own accord. What we are in fact doing here is making the feedback positive, a well-known method of getting a very low source impedance. If we are to do this it must be by some other technique. Suppose that we do use positive feedback and that by some means still rather obscure we maintain the unity ratio from the driver to the transistor currently being driven. The driver cannot provide the necessary drive current, since this is the emitter current, of the driven transistor.

Increased Current Drive

We may well do better to provide more current drive and a form which I find attractive is the arrangement shown in *Fig.* 6. This is a split-load circuit in which $n = 1$, so that the available drive current at one half-secondary is twice the standing current of the driver transistor. The four equal windings can be very tightly coupled, because we can use quadrifilar, twinned-twin, winding. True, the impedance which drives the power transistors will have been ½ ohm instead of 1 ohm, the value we should get with an emitter follower and a trifilar winding, so that we should, so far as I can see, actually gain on the deal.

(Continued on page 65)

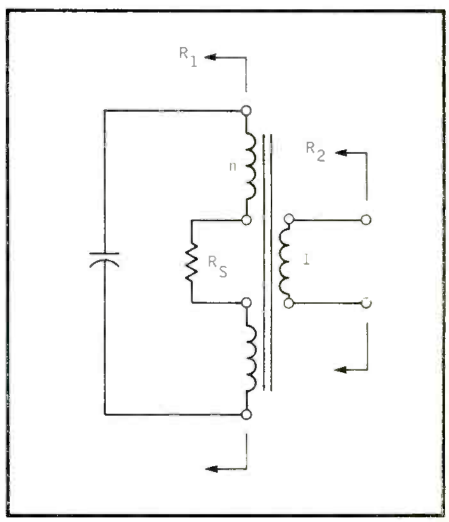

Fig. 7. Calculation of driver impedance.

Putting Junkbox Meters to Work—II

DONALD R. HICKE*

A discussion of the various types of meters for AC and how to convert a "junkbox" d.c. meter to a.c. measurements.

In Part I of this series (August, 1965) the operation of an ordinary d.c. moving-coil meter was explained and the procedure for determining the electrical characteristics of the coil was given. Using these characteristics, design equations were presented to enable the reader to convert the junkbox meter into a useful voltmeter, ammeter, or ohmmeter. All of the material presented, however, dealt with direct-current operation; in this installment the problems of making similar measurements on alternating currents will be outlined and methods given for converting the same junkbox meters to that purpose.

The Problem

For those who may have missed Part I, it was explained that the ordinary d'Arsonval meter consists of a coil of wire mounted on pivots within the air gap of a powerful permanent magnet and that the coil, being restrained with springs, is deflected an amount proportional to the current flowing in it. The direction of rotation depends upon the polarity of the permanent magnet, the direction in which the coil is wound and the direction of current flow in the coil. The first two of these factors is fixed by the meter manufacturer; he makes an attempt to define the third one by marking the meter terminals with "+" and "−". Those who fail to heed these markings are usually rewarded with a muted thud as the pointer slams against the lower stop. Ordinarily there is no damage to the meter when this happens, but notice that it demonstrates the bi-directional property of the permanent-magnet meter.

If the pointer is purposely offset to center scale by adjusting the zero set screw, and if a small current is applied alternately in one direction and then the other, the pointer happily flicks back and forth indicating the applied polarity. Whether or not it also indicates the magnitude of the current depends on how rapidly the current is reversed and how fast the pointer can move. As the frequency of the applied reversals is increased, a point is soon reached where the pointer gives up and refuses to move at all. Since this occurs at only a few tens of Hz at the most, it is readily apparent that some other method must be found to measure a.c. at the frequency most often encountered by the home experimenter—60 Hz.

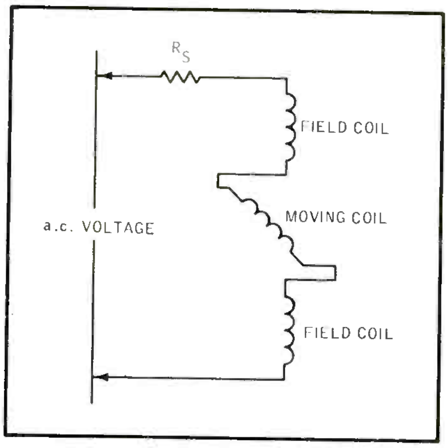

Fig. 2-1. The permanent magnet is replaced by field coils in this a.c. voltmeter, called an "electrodynamometer."

The Solution

One way to solve the problem of the reluctant meter movement is to make the "permanent" magnetic field oscillate at the same rates as the current flowing in the moving coil. This is accomplished in the electrodynamometer by replacing the horseshoe-shaped magnet with large field coils between which the moving coil is mounted, as shown in *Fig.* 2-1. Now the torque acting on the moving coil is always in the same direction, and the pointer can respond. When measuring current, the field coils are made of heavy wire and a shunt is connected across the moving coil, because it cannot carry heavy currents. In the earlier article, it was stated that the deflection of the moving coil is proportional to the strength of the magnetic field and to the current flowing in the moving coil. The same holds true for the electrodynamometer, but since the magnetic field is also dependent upon the current being measured, the deflection of the coil becomes proportional to the square of the current. When measuring a.c., the pointer responds to a value known mathematically as the mean-square of the current and the scale is marked in terms of the square root of this quantity. Thus the electrodynamometer indicates the root-mean-square value of an alternating current, which by definition is equivalent to a direct current of the same magnitude. This will be explained in greater detail later on. Because it responds equally well to d.c. and a.c., it may be used to calibrate other a.c. meters by referencing a known d.c. source. Because of its size, complexity, and precision, the electrodynamometer is a true laboratory instrument and is not likely to be found in the junkbox.

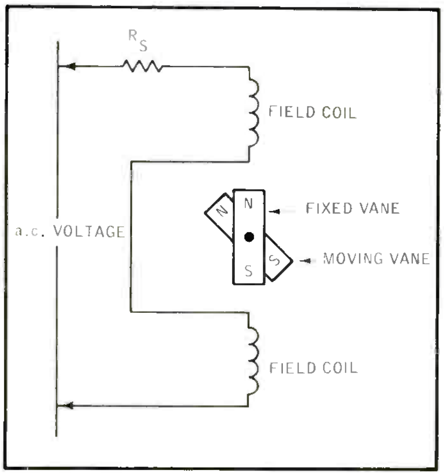

Fig. 2-2. Repulsion of like magnetic poles provides the indicator movement in the moving-vane a.c. meter.

Another approach to the problem of measuring a.c. is to use the field coils to generate an alternating magnetic field, but replacing the moving coil with two pieces of soft iron. Several different types of construction are possible, but the basic idea may be seen by referring to *Fig.* 2-2, which shows the two iron vanes mounted between the coils. Because both vanes are in the same magnetic field, both are magnetized in the same direction; but because like magnetic poles repel each other, the moving vane is repulsed and carries with it a

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pointer to indicate the amount of current flowing in the coils. When the field reverses direction the magnetic poles in the iron vanes are also reversed, but they are still similar to each other and continue to repel. The deflection ordinarily follows the mean-square of the current as in the electrodynamometer, but it can be made nearly linear over most of the scale by properly shaping the vanes. Unfortunately, this meter cannot be used as a reference on d.c. because the vanes tend to become permanently magnetized.

The moving-vane meter is widely used in inexpensive a.c. ammeters, and one or two samples may possibly be found in the junkbox. Changing the full-scale range involves removing the field coil while noting the number of turns, then rewinding it with a greater or lesser number of turns, depending on whether the full-scale range is to be decreased or increased. For example, if twice the sensitivity were desired, twice the number of turns would be needed, and so on. This may require using a smaller diameter wire to get it all in the space available. The author has not tried to modify a moving-vane meter in this manner because, frankly, it sounds like an awful lot of work for a three-dollar instrument.

Rectifier-Type Instruments

Still another solution, and the one that has been the most popular, is to leave the meter alone and instead modify the current flowing through the moving coil. The simplest approach is to use a single rectifier cell, connected as shown in *Fig.* 2-3. The rectifier can be any of the standard types, although germanium and silicon diodes are gaining in popularity over the more commonly-used copper oxide rectifiers. The diode conducts current only in the direction indicated by the arrow, and clips off negative excursions of the applied voltage. An inexpensive 1N34 or 1N58 may be used. In the equation for R_s, use the rms value for voltage and the full-scale meter rating for current, and use the value of diode resistance at peak meter current (more about the diode resistance later).

The moving coil responds to the average value of the current flowing through it, which can be shown mathematically to be equal to 0.318 of the peak value *for a sine wave only*. The meter also indicates the average value of a complex wave form, but it will not necessarily be equal to 0.318 times the peak value. In fact, the half-wave a.c. meter has the useful property of giving an indication of the degree of non-symmetry of a complex waveform by simply reversing the leads and comparing the readings.

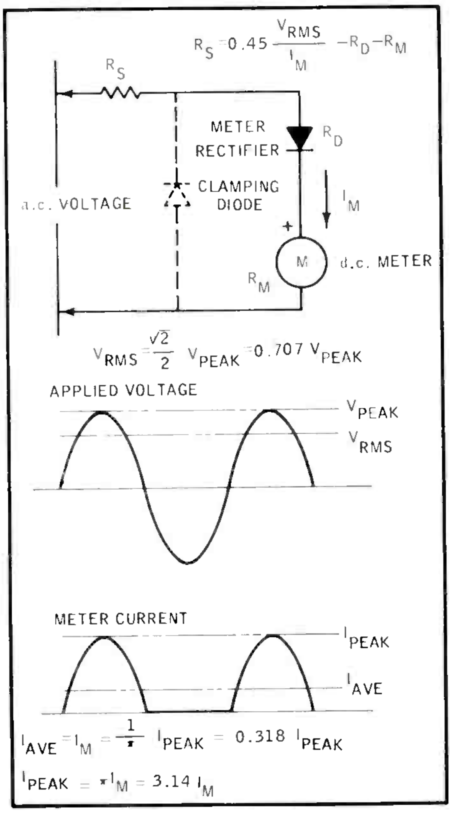

Fig. 2-3. The half-wave rectifier meter circuit. The equations are valid only for sine waves.

Some of the disadvantages of the circuit described are that it is subject to rectifier burnout, requires a non-linear scale, and is not very sensitive. A careful examination of the nature of the rectifier shows why this is so and what can be done about it.

One of the characteristics of a semiconductor rectifier is that it can withstand only a limited amount of voltage in the reverse direction. If this "peak-inverse-voltage" (P.I.V.) is exceeded, the device conducts current in the reverse direction and is no longer a useful rectifier. In a copper oxide rectifier this breakdown point occurs between 2 and 5 volts, but for germanium and silicon it may be as high as several hundred volts. The 1N34 germanium diode, for example, has a P.I.V. rating of 75 volts.[1] Another characteristic is that even at low reverse voltages there is a tendency for the rectifier to conduct a small amount, in the order of microamperes, which subtracts from the forward current and causes the meter indication to be low. The 1N34 is rated for a maximum reverse current of 50 μA at 10 volts and 800 μA at 50 volts.[2]

The dotted lines in *Fig.* 2-3 show a commonly-used solution to both problems: a clamping diode, which may be another 1N34, is connected across the meter and its rectifier. When the applied voltage is in the positive direction, the clamping diode has no effect upon the circuit. In the negative direction, however, it protects the meter rectifier from burnout by conducting and reducing the voltage across the meter rectifier to approximately 0.2 volts. At this low value of reverse voltage, current through the meter rectifier is negligible.

The non-linearity of the circuit is caused by another characteristic of semiconductors. *Fig.* 2-4 shows a plot of voltage drop *vs.* forward current in two germanium diodes. The measurements were made on two units the author happened to have on hand and may not be representative of all diodes of the same type, but the shape of the curves is typical. It shows that the diode resistance varies greatly at low current levels and continues to be non-linear, although to a lesser degree, at higher currents. This is why the scale of most a.c. meters looks crowded at the left side but approaches a linear scale at the right side. From a curve of this type, the diode resistance R_d may be calculated for use in the equations of *Fig.* 2-3.

The low sensitivity of the half-wave meter circuit is caused by clipping off and throwing away half of the applied voltage. An easy way to almost double the sensitivity is to use a full-wave rectifier circuit, as shown in *Fig.* 2-5. Here the negative excursions of the applied voltage are inverted and used, instead of being thrown away. The meter indicates 0.635 times the peak value of an applied sine wave, minus the voltage drop through the two diodes in series. Sometimes, in the interests of economy, two adjacent diodes are replaced with fixed resistors. The value varies from 500 to 5000 ohms depending upon the type of meter, the type of diodes, and the meter sensitivity desired. A higher resistance would tend to swamp out

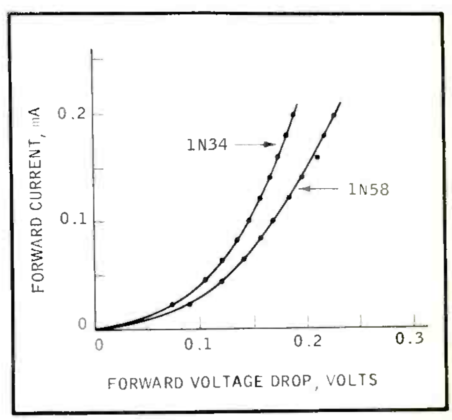

Fig. 2-4. Forward voltage drop across two typical germanium diodes.

[1] "Semiconductors", *General Electric Company*, publication 640.10, page 18.

[2] *Ibid*

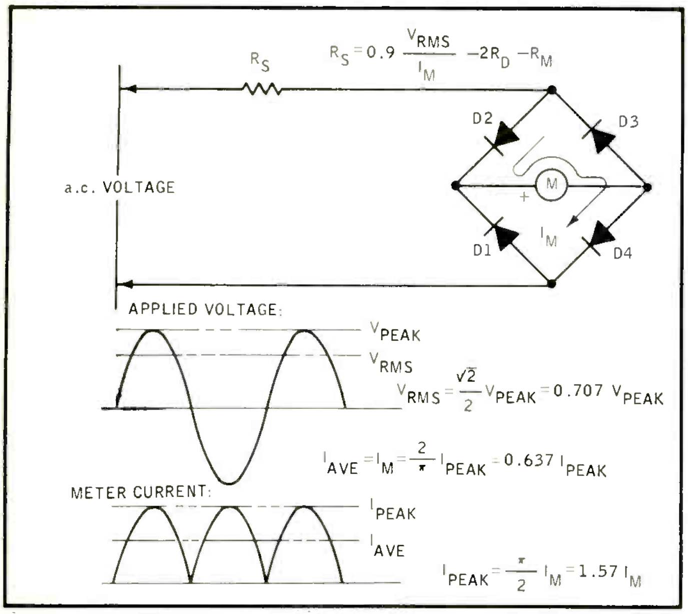

Fig. 2-5. Full-wave meter rectifier circuit. The equations are valid only for sine waves.

variations of the diode resistance and make the meter scale more nearly linear, but it requires a more sensitive meter movement. A lower resistance is required for less sensitive meters, but it cannot be made too low because it is shunted directly across the meter and reduces the resistance (ohms-per-volt) of the entire voltmeter circuit.

Calibration

The calibration of a d.c. voltmeter circuit is usually no problem: one simply obtains a known d.c. voltage source, such as a fresh mercury cell, and tailors the value of series resistance R_s until the meter indicates the proper voltage. Higher ranges are obtained by constructing a voltage divider with precision resistors, and checked with several battery cells connected in series. Unfortunately, calibration of an a.c. meter circuit is not quite as simple. In the first place, there are several ways to describe alternating voltages, and, secondly, they do not come in convenient little packages for calibration, like mercury cells. Both of these problems can be overcome, however, if the nature of an a.c. voltage is understood.

The most easily-recognized description of an alternating voltage is its peak-to-peak amplitude. When viewed on an oscilloscope, it is the total vertical distance from the top of the positive peak to the bottom of the negative peak. This value is important to an audio or communications engineer, because many signals are defined by their peak-to-peak values. A moving-coil meter with a full-wave rectifier and a large capacitor could be used to measure peak voltage, but the meter requires a certain amount of current, typically anywhere from 50 microamps to 20 milliamps. This tends to discharge the capacitor between voltage peaks, causing the meter to indicate low. In addition, signal sources of the type which would require peak readings usually cannot supply the power it would take to operate a moving-coil meter. Generally, peak-to-peak voltage measurements are best left to specialized instruments such as oscilloscopes.

Another obvious characteristic of a symmetrical a.c. voltage is that its average value is zero. That is, it will cause as much current to flow in one direction as it does in the other. But if our junk-box meter requires too much current to measure peak voltage, and if it responds only to the average value of current in the coil, how can we measure an a.c. voltage? The answer is that we cheat—a little. Both the half-wave rectifier circuit of *Fig.* 2-3 and the full-wave circuit of *Fig.* 2-5 allow an a.c. voltage to be measured by distorting it so that it is no longer symmetrical. The half-wave circuit simply throws away the negative excursions; the full-wave circuit inverts them and sandwiches them between the positive peaks. Now the meter has current flowing through it in one direction only, and it indicates the average value of this current.

Unfortunately, the average current from a rectified sine wave is of little significance, because it varies with the type of meter circuit and with the losses in the rectifiers. The ideal solution would be to choose some representation of an a.c. voltage which is independent of the particular method of measurement and which can be easily duplicated by anyone. The standard usually used is the root-mean-square, or rms value, whch was mentioned briefly before. It is defined as that amplitude of alternating current which does the same amount of work as a certain value of direct current. It can be shown mathematically that a sine-wave current of 2.82 units peak-to-peak, or 1.414 units from zero to peak, is equivalent to a direct current of 1.0 units. This suggests that the a.c. meter circuit may be calibrated directly by measuring the ability of the a.c. voltage to produce heat in a resistor and comparing it to a d.c. voltage which produces the same amount of heat. Although this is the right way to calibrate an a.c. meter, the process of taking temperature readings on a resistor is tedious and time-consuming, and subject to errors from changes in ambient temperature or non-repeatability of the thermometer used.

A far simpler method which has sufficient accuracy for the average home experimenter is to assume that the a.c. voltage is a pure sine wave and measure its peak value with a calibrated d.c. meter, then compare this to the average

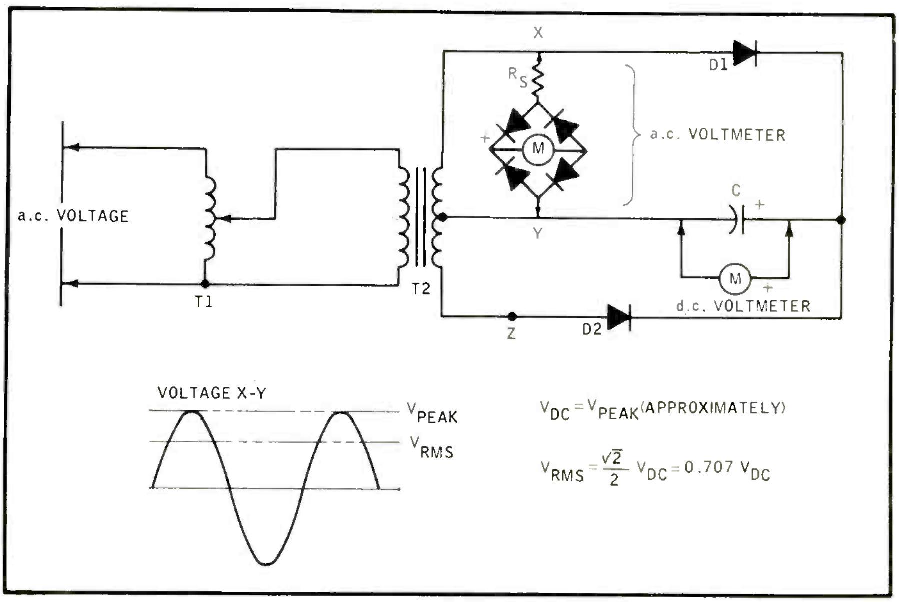

Fig. 2-6. Circuit for calibrating an a.c. voltmeter.

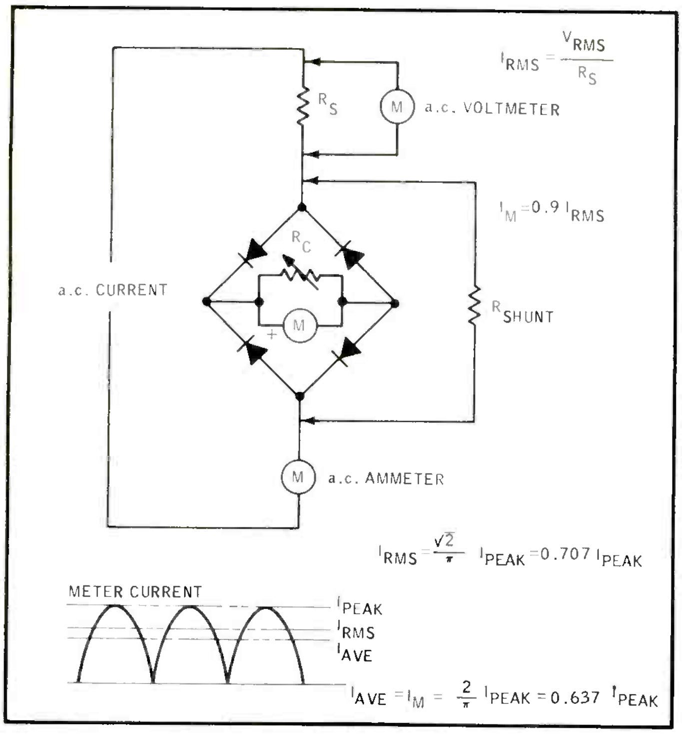

Fig. 2-7. Calibration circuit for an a.c. ammeter.

reading on the a.c. meter. A suggested circuit is shown in *Fig.* 2-6. The two transformers provide a source of variable a.c. voltage. Almost any power transformer may be used for T_2, but it should have a fairly high output voltage to minimize the effect of the voltage drop across the diodes. A model-train transformer could be used for T_1, feeding into a filament winding on T_2. The capacitor should be 30 microfarads or more; select one that gives the highest reading on the d.c. voltmeter, as this signifies low leakage and sufficient capacitance to remain charged between voltage peaks. Be sure not to exceed the working voltage rating of the capacitor. The diodes must be able to withstand a peak inverse voltage of at least twice the d.c. voltmeter indication. From a catalog, select one based upon the rating of transformer T_2 and the highest d.c. voltage expected.

If all this sounds like too much bother and expense, use the power supply in an old amplifier or television set. Remove all the tubes except the rectifier and disconnect all loads from the filter capacitor. This will give a one-point calibration which may be sufficient for some applications (the train-transformer idea could still be used if a separate, fixed voltage is supplied to the filament of the rectifier tube).

The actual process of calibration is quite simple. The output of the variable transformer T_1 is increased until the d.c. voltmeter, which was previously calibrated as explained above, and the a.c. voltmeter to be calibrated both indicate near full scale. Because of the filter capacitor C, the d.c. reading is equal to the peak voltage from the output of transformer T_2. The a.c. meter, although it indicates the average value of the rectified current flowing through it, may be marked in terms of the rms value of the a.c. voltage. Adjust both series resistor R_s and the variable transformer so that the a.c. meter indicates the proper voltage at full scale. For example, if the a.c. meter to be calibrated approaches full scale when the d.c. meter indicates approximately 70 volts, adjust T_1 so the d.c. meter indicates exactly 71.4 volts. Then adjust R_s until the a.c. meter is exactly at full scale and mark the meter 50 volts. Reduce the d.c. voltage in steps, marking each point desired on the scale of the a.c. meter.

To avoid errors in the measurement because of unequal voltages from the two halves of the secondary of transformer T_2, it might be a good idea to connect the a.c. meter to points X and Z, instead of X and Y. The meter will indicate twice as high as before, so simply multiply the calculated values by two when marking the scale.

AC Ammeters

The same meter-rectifier circuits discussed previously may also be adapted to measure alternating current. For low values of current, the meter and its rectifiers are simply inserted in the line. If using a full-wave circuit, select a meter movement whose full scale direct current is approximately 0.9 times the maximum rms current expected. If using a half-wave circuit with a clamping diode, the factor is 0.45. This is to compensate for the fact that the meter responds to the average value instead of the root-mean-square value of the alternating current.

Higher currents require the use of a shunt or transformer. In either case the object is to produce a sufficient voltage drop across the shunt or to generate a large enough voltage in the transformer to give usable deflection of the meter movement.

Calibration of an a.c. ammeter may be made using the circuit of *Fig.* 2-7. The test current is checked either with another ammeter of known accuracy or by measuring the voltage drop across a known resistor with an a.c. voltmeter. Calibrating resistor R_c is adjusted so that the meter indicates the proper full-scale current. Use a potentiometer of about five times the meter resistance. If a shunt is used for heavier currents, it may be adjusted instead of resistor R_c. Then the current is reduced in even steps, and the proper value marked on the meter scale.

VU Meters

Some readers may have had occasion to monitor audio signal levels in a tape recorder and been disappointed in the performance of the usual neon bulb in-

(*Continued on page 62*)

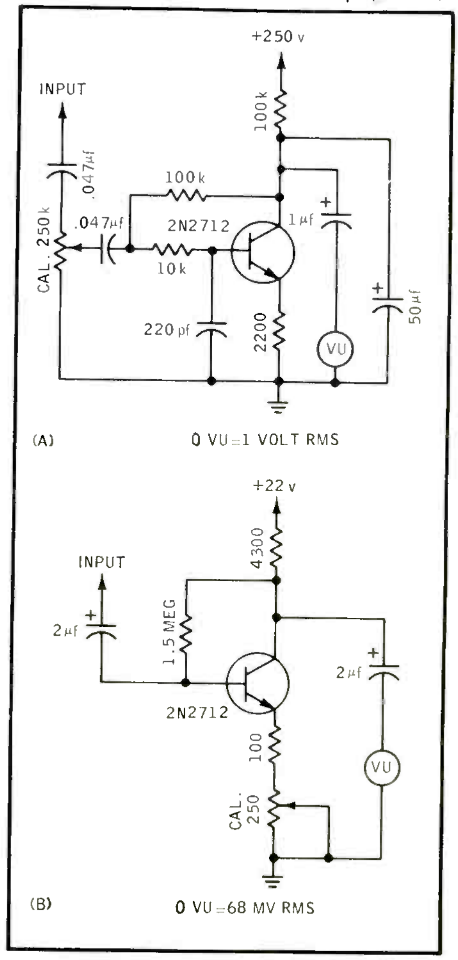

Fig. 2-8. Two VU-meter circuits which may be added to a tape recorder. (A), for a "0" level of 1 volt, and (B) for a "0" level of 68 millivolts.

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Features and Specifications

Tuner Section:
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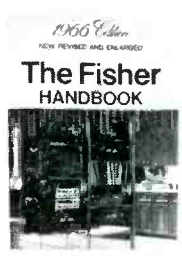
Amplifier Section:
Silicon output transistors; short circuit protection; speaker selector switch (main or aux.); front-panel headphone jack; music power (IHF), 4-ohms, 70 watts; harmonic and IM distortion, 0.8%; frequency response (overall), 20-22,000 cps ± 1.5 db; hum and noise, 80 db; input sensitivity, phono magnetic (low), 4.5 mv; stereo separation, phono magnetic, 50 db.

Size: 16¾" wide x 5⅛" high x 12⅜" deep (including knobs and heat sink).
Weight: 21 pounds.

*PATENT PENDING



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NORMAN H. CROWHURST*

In this first installment of a new Series by this author, a thorough outline is given of the things we must measure in order to assess the performance of an amplifier. The types of tests are described, together with an outline of what measurements we must make, and the equipment set-up required to make them.

Technically, the term "audio" applies to any signal in the audible frequency range which may or may not have been converted from a sound wave, or which may be used to produce a sound wave. So the basic piece of equipment to handle audio is an amplifier, and the measurements one needs to make on it concern the fidelity with which it handles the audio frequencies presented to it.

Parameters to Measure

The easiest way to comprehend the relationship of the various parameters is to discuss them in the order their importance was first realised. First was:

Frequency Response. Any musical or audible program contains a variety of frequencies, from the lowest audible up to the highest audible. The commonly accepted range is from 20 Hz, periods, or vibrations per second, up to 20,000 Hz, or 20 kHz. Actually, many people cannot hear a pure tone of 20 Hz, unless it is loud enough to generate harmonics in the middle ear, in which case its presence is identified by these harmonics; and few people can hear 20 kHz at the other end. Most hearing facilities "die" somewhere in the region from 10 to 18 kHz. But the range of 20 Hz to 20 kHz uses nice round numbers, and ensures that the performance is adequate, in this regard, for the most sensitive human ears.

*Box 651, Gold Beach, Ore. 97444

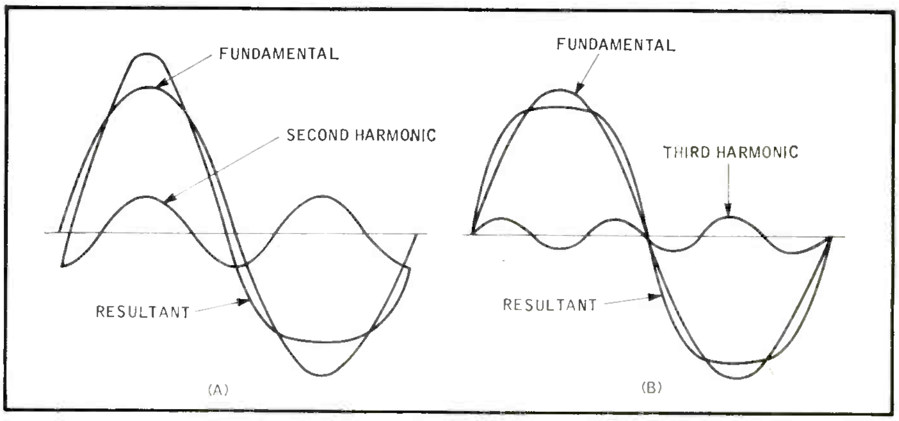

Fig. 1-2. How an added harmonic can change a waveform: (A) second harmonic; (B) third harmonic.

When frequency response was first prescribed as a measure of fidelity, the assumption was that "quality" of reproduction suffered due to some of the original frequencies getting lost, or being relatively reduced in magnitude. As reproducers of the time were predominantly of the moving-iron variety, and many were still using acoustic phonographs, while the telephone concentrated its response close to 1000 Hz, this was undoubtedly the most obvious defect in reproduction at that time (*Fig.* 1-1). Deficiency in response to frequencies away from a rather strident mid-frequency resonance lent considerable coloration to the reproduction.

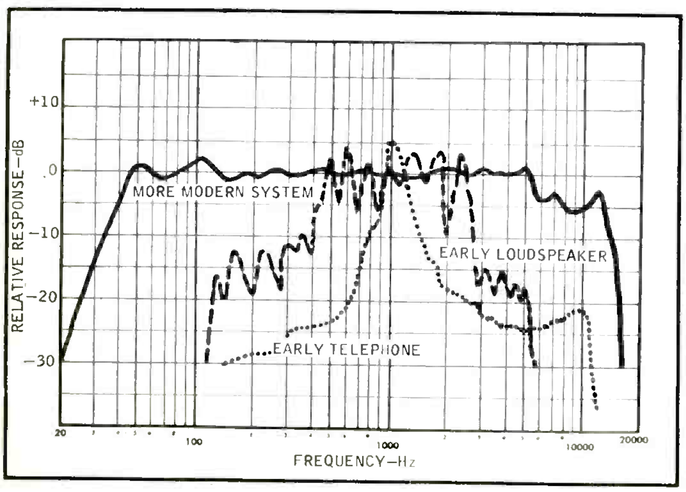

Fig. 1-1. Frequency response curves of several early devices, compared with a more modern system, to show why frequency content was the first criterion of quality considered.

But as amplifiers and reproducers—transducers—were developed with more and more adequate frequency response, it became evident that specification of frequency response did not tell everything. Tests with amplifiers and calibrated attenuators showed that human hearing can not perceive change in level less than 1 decibel, which represents a voltage or current ratio of 12 per cent up or 11 percent down. As equipment could soon be produced that would stay within these limits from 20 Hz to 20 kHz, which still sounded audibly defective, this led to the conclusion that some other kind of measurement was necessary. This led to the recognition of:

Harmonic Distortion. The major notion in measuring frequency response was that some frequencies present in the original were missing, or were relatively reduced in magnitude. The recognition of harmonic distortion was the first measurement directed at finding frequencies that should not be there, something added by the equipment. Due to non-linearity of the amplifying or other handling function, a sinusoidal input waveform departed from its true shape, and the resulting output wave could be analyzed into the fundamental (a pure sine wave) and spurious harmonics (*Fig.* 1-2).

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For years since its introduction, the Dynaco preamplifier design has been generally accepted as one in which the noise and distortion are so low and the quality so high that attempts to improve it would be laboratory exercises rather than commercial enterprises. Yet we have always been questioned as to why we did not gild this lily by adding step type tone controls. The enthusiastic audiophiles who ask this tell us that they want to be sure that their tone controls are out of the circuit when not being used. Our answer has always been that continuous controls give a range of flexibility which cannot be attained with step type controls, and that the "neutral" position of our controls produces a flat response characteristic adequate for the most critical need.

However, our avowed philosophy of perfectionism has kept us working on the possibility of some improvement in the circuit—and this work has now led to the first major change in our preamplifier design since it was initiated. This development (on which patents are pending) is applicable to all continuous tone control systems and immediately makes them superior to the far more expensive step type controls. What we have accomplished is to keep the infinite resolution capability of the continuous control, but to remove all **frequency and phase discriminating networks from the circuit** when the control is rotated to its mechanical center. This new design is incorporated in the PAS-3X (PAS-2X, too) which is now at your dealer's at the same low price.

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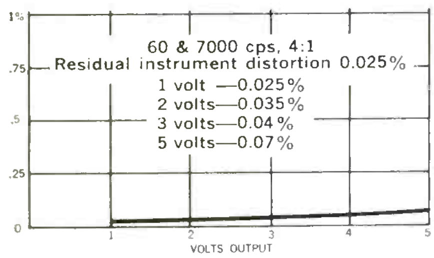
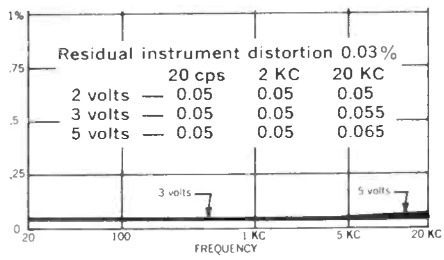
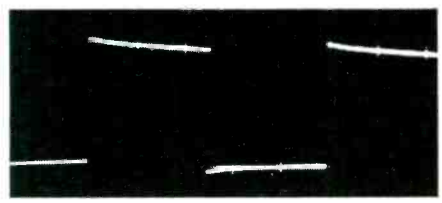
Can you hear the difference? We doubt it. The preamp was amazingly good in the past. We have improved it for the sake of improvement, not because we think it needed it. It has always surpassed every other preamp without regard to cost. And, it is superior on more than measurements—listening tests prove that the Dyna preamp adds no coloration to the sound and that its inclusion in the hi fi chain is undetectable. Partially diagrammed below is the performance you can expect from the PAS-3X—why you can never get better overall quality regardless of how much money you spend.



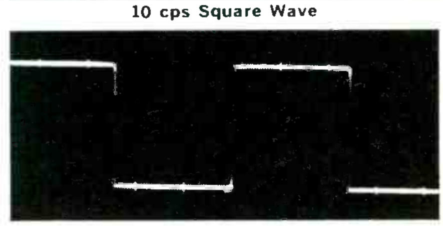
Intermodulation Distortion



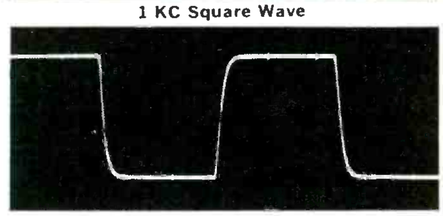
Harmonic Distortion



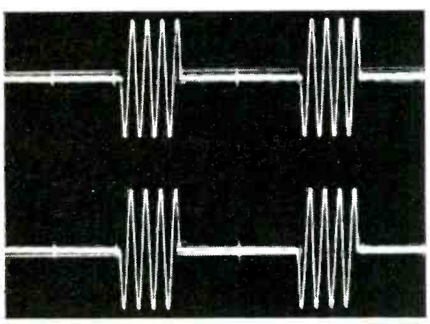
10 cps Square Wave



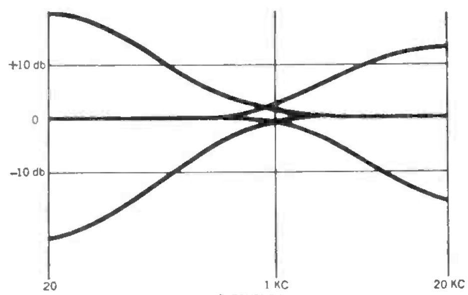
1 KC Square Wave



10 KC Square Wave



Four cycle 20KC tone burst from generator (above) matches PAS-3X (below)



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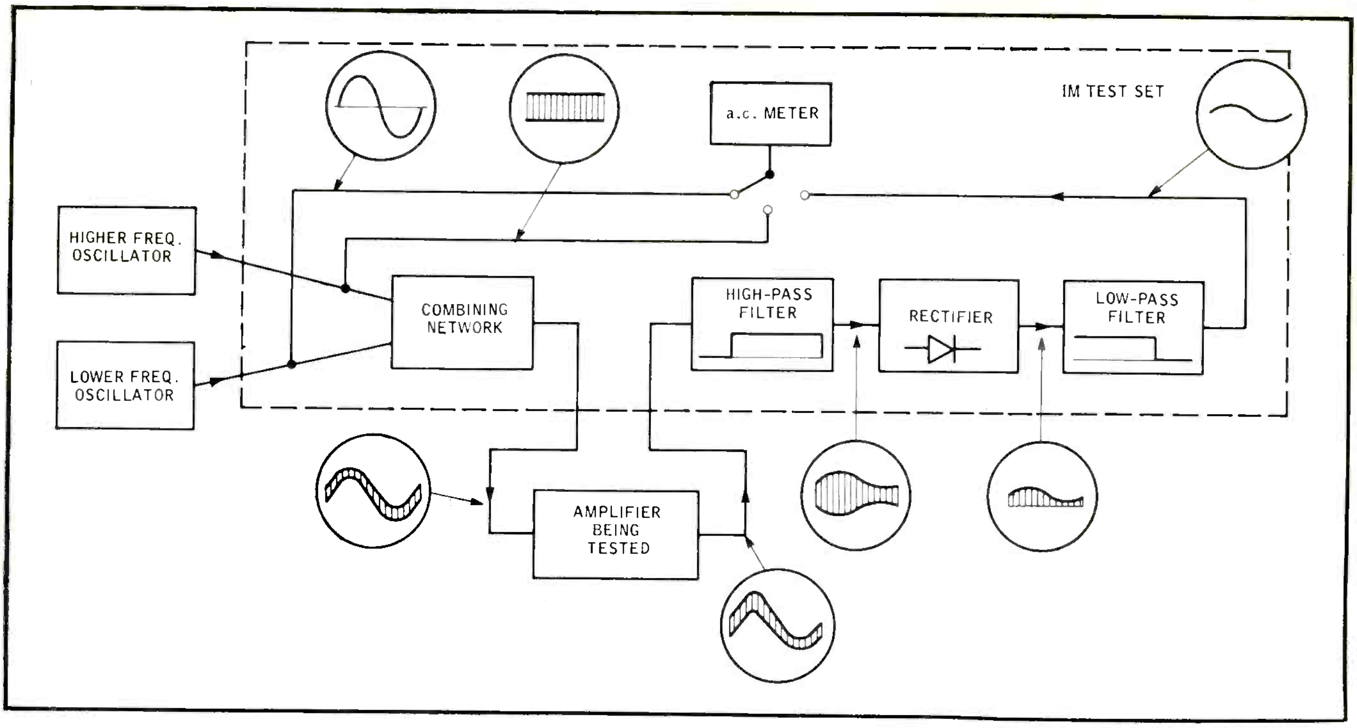


Fig. 1-3. The complete set-up for making the SMPE intermodulation test

Next thing, of course, as with frequency-response measurement, was to find out what relative importance the distortion had. Tests at that time, mainly using second harmonic (first overtone in music) showed that few listeners could detect less than 5 percent added harmonic, as different from pure fundamental. Some researchers went further, checking the effect of harmonics higher than second, and found that, the higher the order of the harmonic, the smaller the quantity needed to make it identifiable as distortion.

But when harmonic measurements had been used as the next means of upgrading equipment, it was found that distortion which should not be at all audible, according to carefully conducted tests, still could be quite noticeable on reproduced music or other program material. This led to recognition of another effect of non-linear distortion, more readily discernible to human hearing than the harmonic form, given the name:

Intermodulation Distortion. For this kind to occur, at least two frequencies must be present at the same time. Both frequency response and harmonic distortion were measured with only one frequency at a time. Frequencies throughout the audible range were used, but only one at a time. This newly recognized form of distortion occurs when the program contains at least two frequencies. Of course, most programs contains many more than two frequencies at a given instant, but to keep measurements simple, only two were considered at one time.

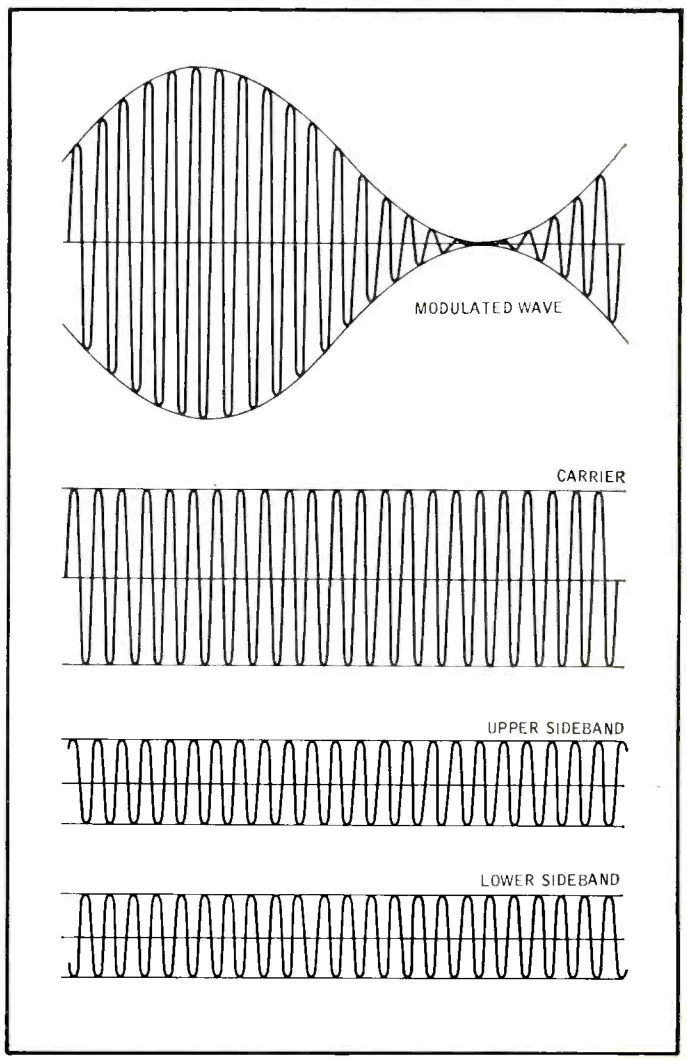
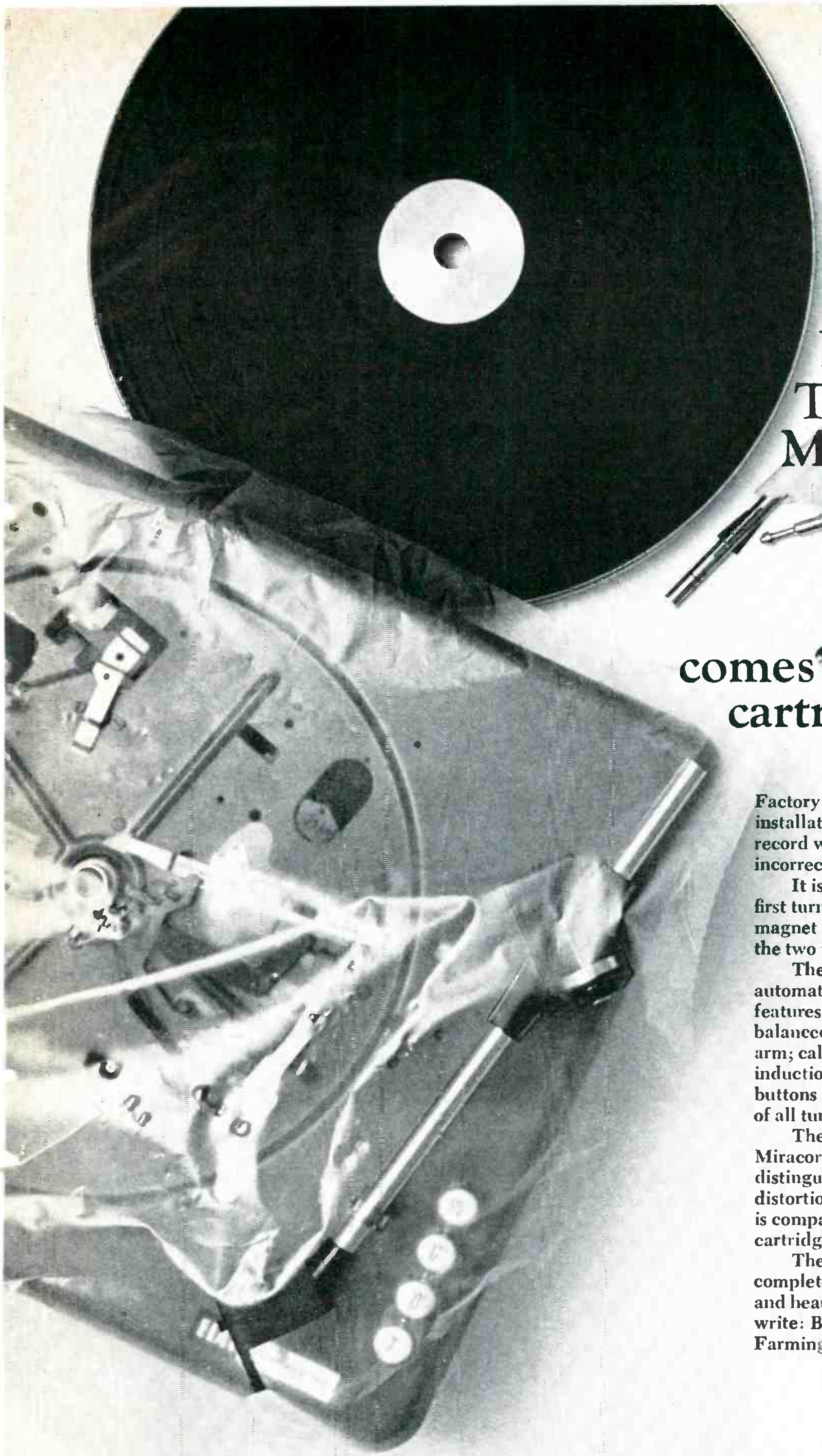


Fig. 1-4. Analysis of an amplitude-modulated carrier, into unmodulated carrier and sidebands. Note that adding the two sidebands together in correct phase or timing relationship, results in the carrier amplitude changing, while its frequency or period (the spacing between points at which it crosses the zero line) doesn't.



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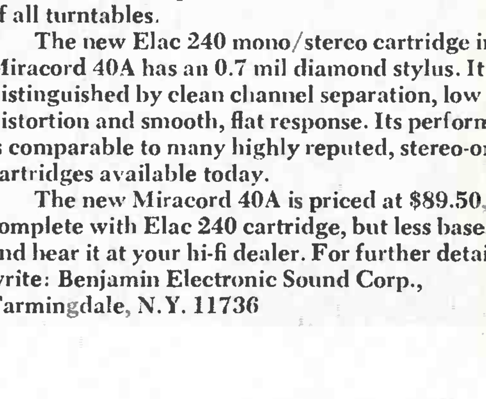

new Miracord 40A

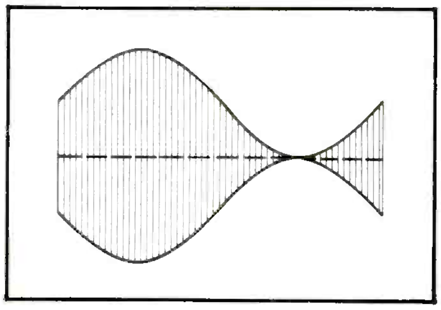

Fig. 1-5. A mixture of two higher frequencies of equal amplitude produces this envelope, which is very similar to the modulated one of Fig. 1-4.

Intermodulation can show up in one of two ways, according to the relationship between the frequencies that give rise to it. If low frequencies and high frequencies occur together (at least one of each) then the low frequencies may modulate the higher frequencies, giving the latter a "dithery" or "gargling" effect. This effect was particularly noticeable on some of the early movie sound tracks, so the Society of Motion Picture Engineers (SMPE, now the Society of Motion Picture and Television Engineers, SMPTE) devised the test that became known as the SMPE intermodulation measurement.

This measurement combines a low frequency and a high frequency in specified amplitude ratio, usually the low one having four times the magnitude of the high one; passes this combination through the amplifier or other part of a system and then measures the modulation of the high by the low by: (1) removing the low frequency; (2) rectifying and filtering the remaining high frequency (*Fig.* 1-3).

For the kind of distortion it aims to detect, that method of measurement seems foolproof at first sight. But the fact is that some quite audible forms can "get by" without detection by this method of measurement. We'll go more fully into that when we describe the method in detail.

The other form of intermodulation distortion is more noticeable when there are at least two higher frequencies, not harmonically related (that is, not in a simple numerical ratio). When the distortion occurs, a lower tone, or buzz, usually betrays its presence. What happens can best be understood by thinking of the usual form of radio modulation in reverse.

When a radio or other higher frequency is amplitude-modulated by a lower frequency, the result can be analyzed into an addition (modulating is really a multiplying of one frequency by another) of two or more higher frequencies (*Fig.* 1-4). So the envelope formed by combining two higher frequencies has a contour that is, or should be, fluctuating at their difference frequency (*Fig.* 1-5). If one frequency differs from the other by 100 Hz (e.g. one is 5000 Hz and the other 5100 Hz) then the envelope will be a wave with a frequency of 100 Hz.

When this form of intermodulation distortion occurs, the envelope is no longer a symmetrical and sinusoidal 100 Hz, but contains some distortion. Usually the sine wave at the top will be bigger or smaller than that at the bottom (*Fig.* 1-6). This results in an actual 100 Hz being produced (heavy dashed line) as well as the *effective* 100 Hz in the modulation envelope. So the 100 Hz now becomes audible, where it would not be, if the modulation wave form were pure, due to the wave only containing the original frequencies (such as 5000 and 5100 Hz).

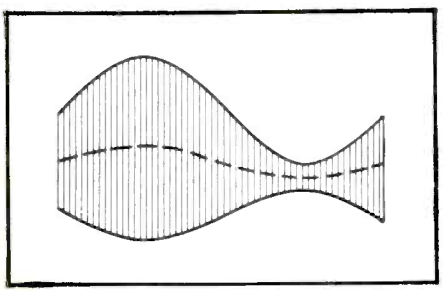

Fig. 1-6. Distortion in an amplifier or system can change the waveform of Fig. 1-5 to something like this one, which includes a lower-frequency component (heavy dashed line) not present in the original wave.

This form of distortion forms the basis of the internationally recognized CCIF measurement (those letters stand for the French name of the international organizations that established the method). It consists simply of using for input any two higher frequencies that differ by a specific number of Hz, usually 100 or 400. The output is then passed through a filter to find whether any 100-Hz (or 400-Hz) tone is present and how much (*Fig.* 1-7). No distortion means no 100 (or 400) Hz present in the output.

In practice, it may be simpler to set one frequency to, say 5000 Hz and then set the other one to approximately 5100 Hz, adjusting it carefully to get a maximum reading on the output from the 100-Hz filter. If a maximum is found closer to the original frequency, such as 5,050 Hz, the reading is due to higher order effects. The procedure for calibrating we will describe when we deal with this method of measurement fully.

This measurement too is of limited validity. But it finds a form of intermodulation distortion different from the SMPE test. Which is the better test has been a subject for much argument. The true answer seems to rest with the purpose of the test, or the kind of equipment on which it is performed. However, neither method is completely satisfactory in "finding" all of the distortion which it is intended to detect. This too we will explain more fully when we describe the tests in detail.

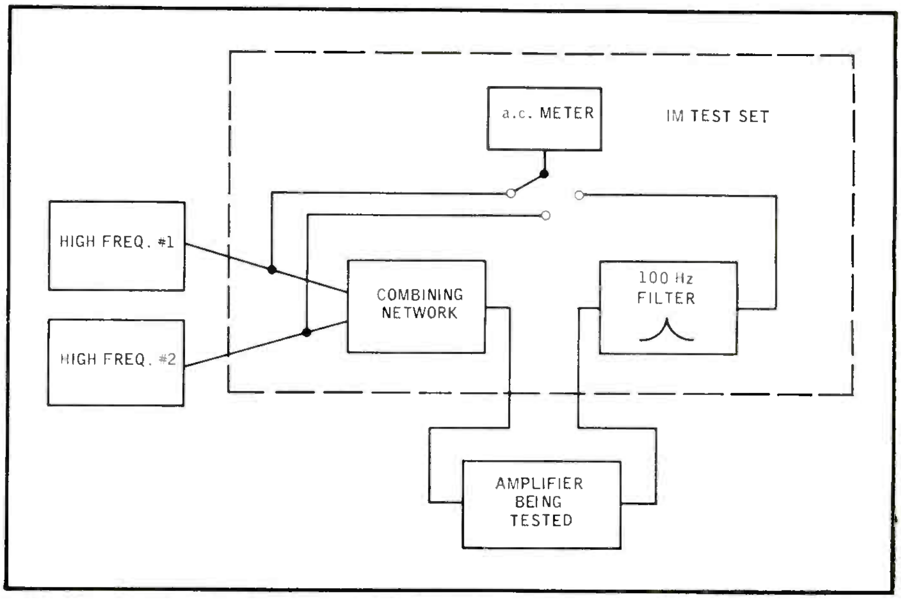

Fig. 1-7. The complete set-up for making the CCIF intermodulation test.

Meanwhile there are two more measurements, one of which began to be important much sooner than the other, that relate to another property of any program material, its

Dynamic Range. This refers to the loudness, or decibel range, between the loudest and quietest sounds in a composite program, or that an amplifier or system can handle. The loudest part is limited in an amplifier or system by the maximum power, while the quietest part is limited by background hiss, rum-

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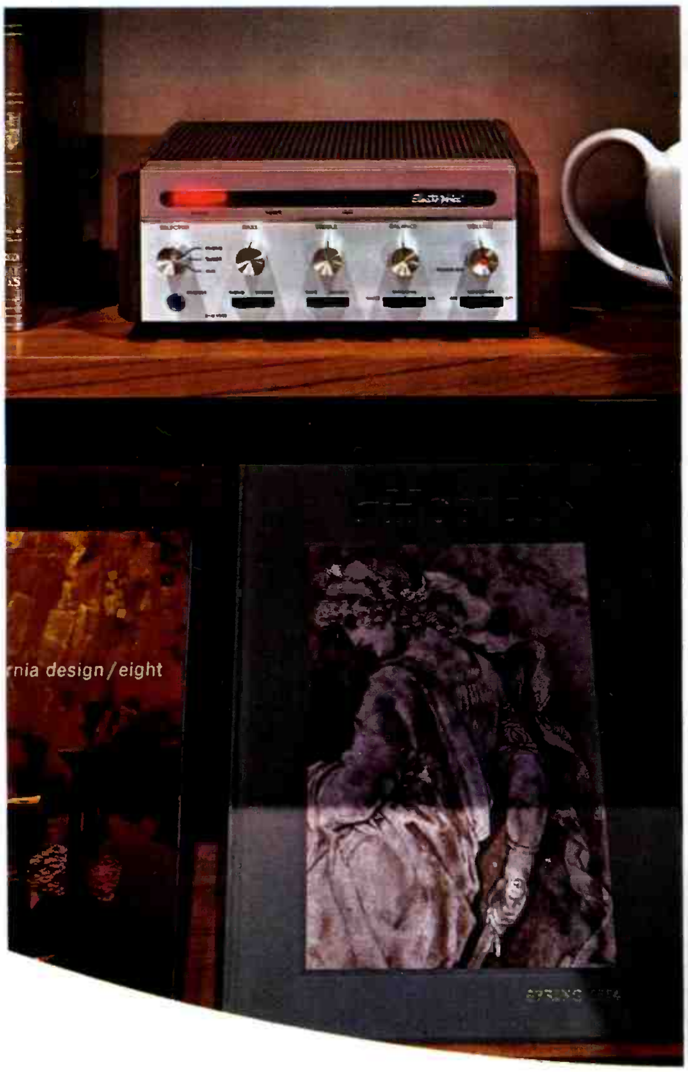

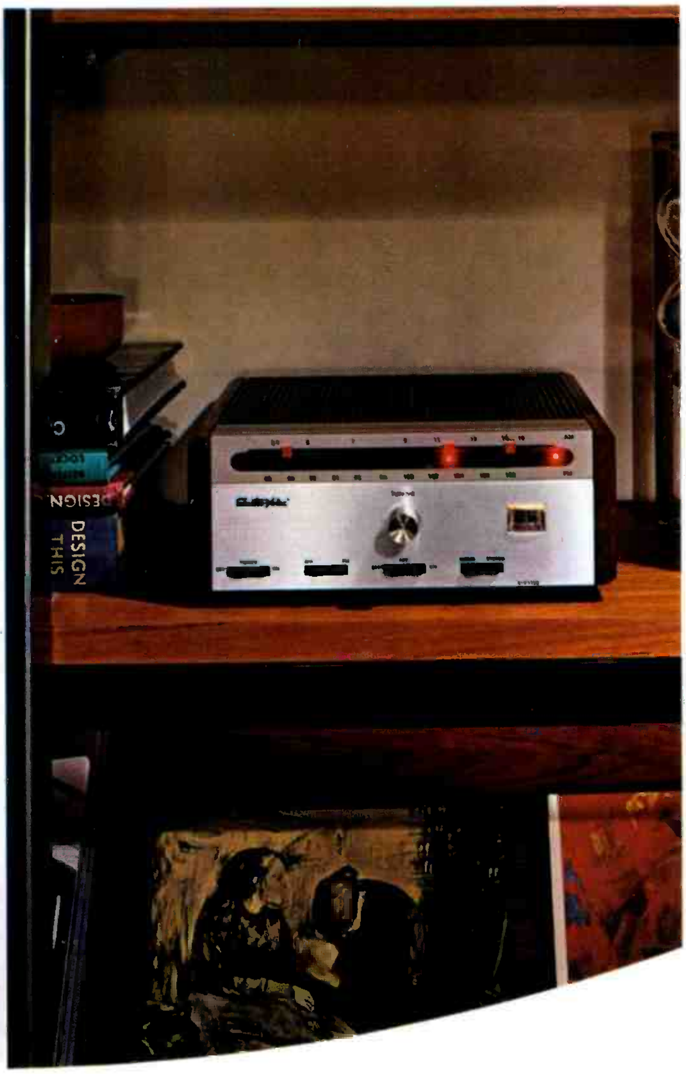

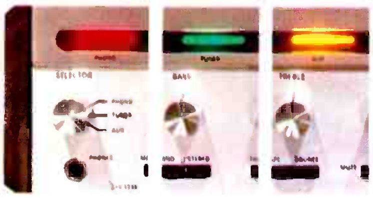

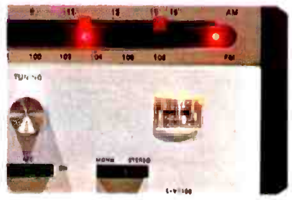

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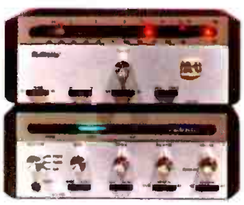

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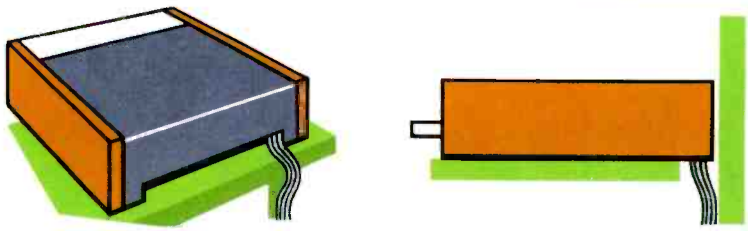

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amplifier specifications / IHF music power output, 50 watts (into 8 ohms, output increases into lower impedances), 25 watts per channel **/ Continuous sine wave output,** 18 watts per channel **/ Frequency response,** ±1.5 db 20-20,000 cps at rated output, ±1.5 db 20-30,000 cps at 1 watt **/ Harmonic Distortion,** less than 1.0% at rated output **/ Hum and noise,** better than 70 db below rated output (magnetic phono input better than 60 db below rated output) **/ Inputs**—phono (mag), tuner, aux, tape (high level). **tuner specifications: Sensitivity,** 2 uv IHF. **amplifier and tuner dimensions:** $3\frac{3}{8}$" high, $8\frac{3}{8}$" wide, $10\frac{1}{4}$" deep. **/ receiver dimensions:** $3\frac{3}{8}$" high, $15\frac{7}{8}$" wide, $10\frac{1}{4}$" deep.

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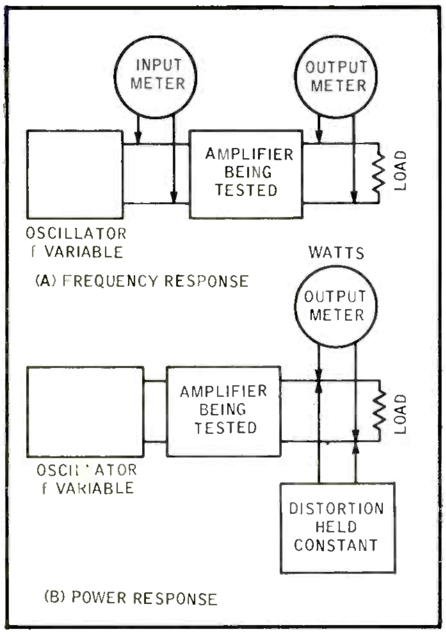

Fig. 1-8. Distinction between method of making (A) frequency-response and (B) power-response measurements.

ble, or hum, collectively called "noise." So dynamic range is not one measurement, but two.

Maximum Power. For the simplest kind of signal, a single tone, like that used for measuring frequency response and harmonic distortion, this is related to both of these forms of measurement. Maximum power will usually vary according to the frequency at which it is measured, and according to the degree of distortion allowed in making the measurement. If a specific degree of distortion is set, lower powers almost invariably result in lower distortion. The maximum power at a particular frequency is the level at which distortion reaches the allowable amount.

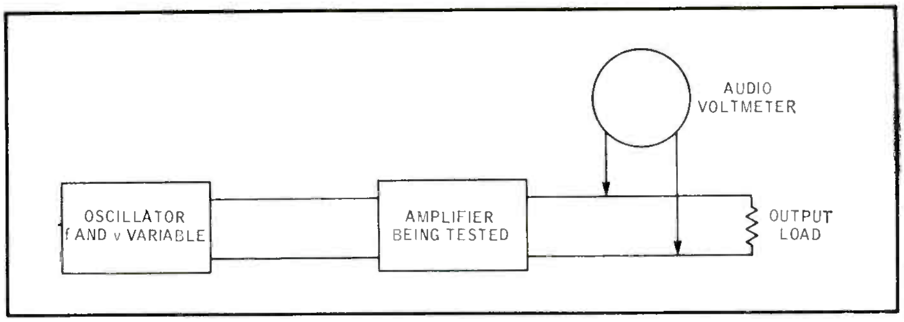

Fig. 1-11. Or is this right? (Answer next month).

Power response relates the maximum power to frequency and reads much like a frequency response, but there is an important difference (*Fig.* 1-8).

Frequency response tells how the output will change as input frequency is changed, provided the output is always within, preferably well within, the maximum power at all frequencies. It assumes constant input level of varying frequency and undistorted output at all frequencies. The measurement may not actually be made at constant input level for all frequencies, as we shall later see, but this is the assumption behind the concept of frequency response. If, to make the measurement, input level has to be changed to avoid distortion at some frequencies, or noise at others, the result is "corrected" to compensate for the change of level actually used.

Power response is not concerned with the input level required at the various frequencies to obtain maximum power, only with how much output power can be obtained at various frequencies, before the distortion limit is reached. Power is the top end of the dynamic-range figure for any particular frequency; the bottom limit is:

Noise. In electronic equipment, such as amplifiers, most background noise is thermal in origin—due to random agitation of molecules in resistors, tubes, transistors, and other components, because of their natural heat—temperature above absolute zero. Such random noise can more accurately be regarded as containing an infinitely random series of events distributed through time, than of frequencies distributed through the audio spectrum (from 20 to 20,000 Hz.). But they are heard as hiss, with various colorations, according to what is, in effect, a frequency content of these random "happenings." So this is the way noise is most often analyzed for audio measurement purposes.

But the level measured is quite different from that used for periodic waveforms, because the noise is not an infinite composure of different frequencies, but an infinite variety of magnitudes and time spacings of pulses, unrelated to one another.

Another element of noise is periodic in nature: hum that somehow gets injected from the supply source (where a system is line operated, or operates in the vicinity of an a.c. power supply or system) or rumble, due to some mechanical action injecting an electrical "signal."

Finally, even assuming that frequency response, harmonic distortion and intermodulation tests, as well as measurements to determine dynamic range, do tell the whole story in their respective areas, certain distortions began to be noticed which quite obviously would not be detected by any of these methods, even if those measurements could be improved to "catch" what they might presently be missing. This deficiency is

(*Continued on page 67*)

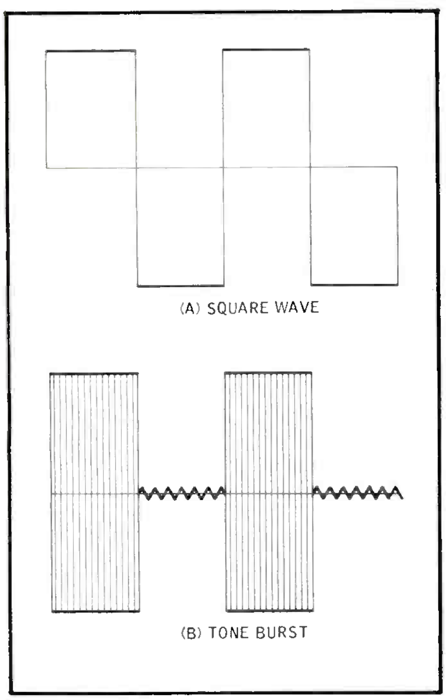

Fig. 1-9. Two types of "transient" test waveform that have enjoyed some popularity.

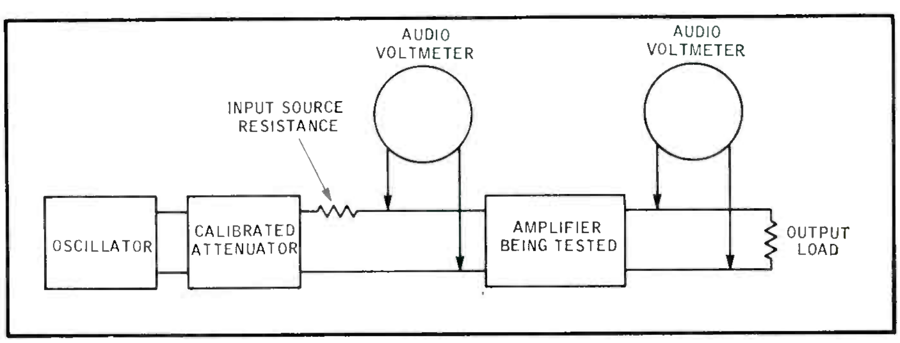

Fig. 1-10. Is this what you need to measure frequency response?

An Improved Omnidirectional Microphone

HAROLD S. MAWBY*

The descripion of the design of a commercial product is often the best source of tutorial information about the device which can be found, particularly when the product has the reputation long enjoyed by the 635 microphone. The design of the 635A, its successor, indicates why it, too, should achieve the same reputation.

THE SIMPLEST AND OLDEST type of microphone, the omnidirectional, has many characteristics which can be put into balance to result in a superior unit when compared to the stereotyped omnidirectional microphone.

While directional microphones are more commonly used today because of the increase of users' knowledge and their own improved quality, there still remains a broad area for the omnidirectional microphone as a necessary and useful audio instrument. This device, being inherently a simpler instrument in mechanical and acoustical design and construction, has been the first in use and the lowest in equivalent cost. It has come through many years of trial and error to result in the high-quality units available on today's market.

Microphone Engineer, Electro-Voice, Inc., Buchanan, Michigan.

For years of dependable service without maintenance of any kind, the Electro-Voice Model 635 omnidirectional dynamic microphone would have a high rating. With the changing times, the studio and station user has had a change in requirements of an omnidirectional microphone. The up-dating of the Model 635 has resulted in the recently released Model 635A, shown in *Fig.* 1.

A comparison of *Fig.* 1 with *Fig.* 2, the older 635, will show that there is no resemblance. Most important, of course, are the changes on the inside of this new shape.

All of the past good qualities of the 635 are the foundation for the many small but important improvements found in the 635A.

Fig. 1. The new Electro - Voice 635A omnidirectional microphone, on a 310 stand clamp. This is an updated version of the "old reliable" 635.

Description of the 635A

Rugged construction was a basic requirement placed high on the list in design. The case is constructed from steel. The front grille work is formed from heavy-gauge wire screening. These result in a durable shell around the intricate transducer.

The generating element is a dynamic or moving-coil unit. The ⅞-inch diameter diaphragm is made of .0015-in. Acoustalloy® which is stable under severe sound pressures and temperatures. The finish has kept in mind the television user requirement of minimum light reflection. A non-reflecting matte satin chrome finish which will withstand much scuffing is used.

The 635A, designed for hand-held use, requires that mechanical shock noise be reduced to a minimum. This consists mainly of contact noise from the case and cable. Two design considerations that contribute much are mechanical isolation of the generating element from the microphone case and the control over the compliance and damping of the dynamic element.

Isolation of the generating element has the disadvantage of enlarging the over-all diameter of the final microphone shape. After many tests with materials of varying compliances and thicknesses, it was found that a simple boot of low-durometer plastisol around the generating element gave necessary isolation with a minimum enlargement of the final shape.

The second consideration is to minimize the effect of the vibrations that are transmitted through the plastisol isolator to the generating element that will put the magnetic structure into vibration. Unless the diaphragm-coil assembly moves with the magnetic structure, there will be a relative motion between the magnetic field and the diaphragm coil, which will result in unwanted electrical noise.

Jensen Model HS-2 Stereo Headphones bring an exciting new dimension to your favorite recordings. Unusually smooth full range undistorted output set these headphones apart. Soft foam cushions provide feather weight comfort and preserve the undistorted bass response without pressure. Response is true, complete, distortion-free from below 20 to 17,000 cycles!

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The analogue of the mechanical and acoustical network can be shown in the electrical form as in *Fig.* 3.

Observing the equivalent network, the velocity of the diaphragm-coil and velocity of the magnetic structure can be made equal if impedance of C_D is large enough. Thus the diaphragm stiffness will help in noise reduction.

If the velocity of the magnetic structure can be made a minimum so that V_M approaches 0, minimum noise will be transferred. This means making the impedance M_m large, which is the mass of the magnetic structure. Desired size and weight of the final microphone must be kept in mind at this point.

Observe also that R_A, the acoustical damping of the diaphragm, must be large enough to control the resonance of M_D with C_D, which is the diaphragm resonance.

Fig. 2. The original E-V 635 microphone.

In selecting the above component values, we must remember the 635A is a dynamic or moving-coil transducer. This requires, from Eq. (*1*), that the velocity v_D of the diaphragm coil be constant for all frequencies over which a constant or "flat" output is desired.

$$e = B\,l\,v_D \quad (1)$$

where e = generated voltage
B = flux density
l = length of the coil wire in the magnetic field
v_D = velocity of the diaphragm-coil wire

The velocity of the diaphragm coil is determined by Eq. (2).

$$v_D = (P/Z) \sin \omega t \quad (2)$$

where v_D = velocity of the diaphragm coil
P = sound pressure actuating the diaphragm
$\int Z$ = Impedance of the mechanical and acoustic system of the microphone

As can be seen from Eq. (*2*), if the pressure P is constant with frequency, then to make v_D constant, the impedance Z must not change with frequency or Z must become R. This is called a resistance-controlled element. This means the diaphragm is highly damped and as discussed previously, is part of the answer to the 635A being less sensitive to mechanical shock. Furthermore, the highly damped diaphragm does not vibrate easily when the performer snaps the microphone in or out of the stand clamp during a performance. If the microphone is being passed from hand-to-hand while "alive," there is little mechanical noise. The highly damped diaphragm is also the fundamental step in making the microphone less susceptable to wind and breath blasts.

The design shape allows the 635A to be used as a hand-held microphone. Thus, there is the user who will use it very close to the lips. This requires protection against "pops" as in a phrase such as "Peter Piper Picked a Peck of Pickled Peppers," which, it has been said, rarely occurs in ordinary conversation. The sudden high sound pressures formed at the lips will require a "pop" or breath-blast filter. This acoustic filter on the 635A is placed in front of the diaphragm directly behind the front grille. This eliminates the need of an external windscreen in most outdoor applications. Just below the "pop" filter is a dust filter.

Between it and the diaghragm is a fine screen constructed of a magnetic material. This screen is in the leakage magnetic field of the generating element. The screen becomes magnetized enough to attract any fine metal dust and filings. This prevents a gradual dirt build-up on the diaphragm, which would cause a change in frequency response, the incidence of distortion, and loss in sensitivity.

Case Considerations

The size and shape of the case is determined by several acoustic necessities and desired user features.

The case diameter of ¾ in. was based around the XLR-3-11 plug. This plug is preferred in the broadcast and recording business.

The ¾-in. diameter is also small enough so that a female performer can handle the microphone comfortably.

A larger diameter at the front of the microphone was based around a generating element which had an output sensitivity that was desirable in a hand-held microphone. The larger the area of the diaphragm, the more coupling there is between it and the air.

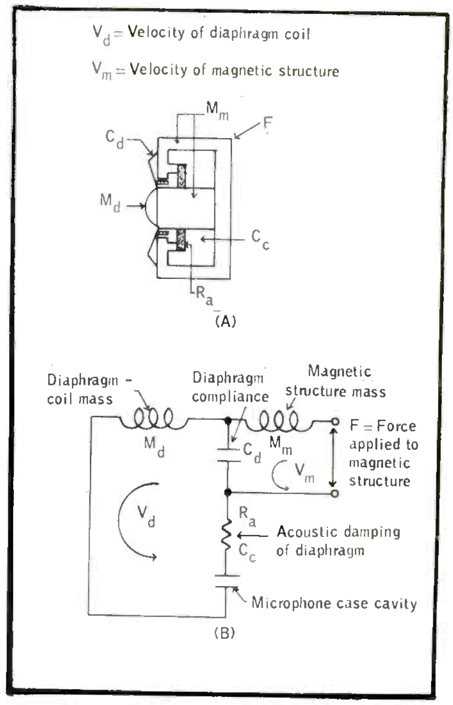


Fig. 3. Mechanical noise isolation in the 635A. (A) cross section of the generating element, and (B) the equivalent electrical network.

Diaphragm area is analogous to turns ratio in a transformer impedance-matching problem. In this case, a larger diaphragm requires a larger terminating sealed cavity behind the diaphragm. The size of this sealed cavity is a determining factor in the support of a low-frequency response.

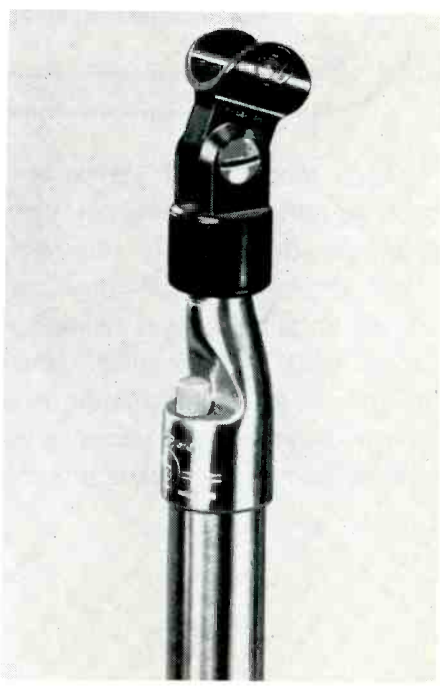


Fig. 4. The "snap-out" Model 311 stand clamp.

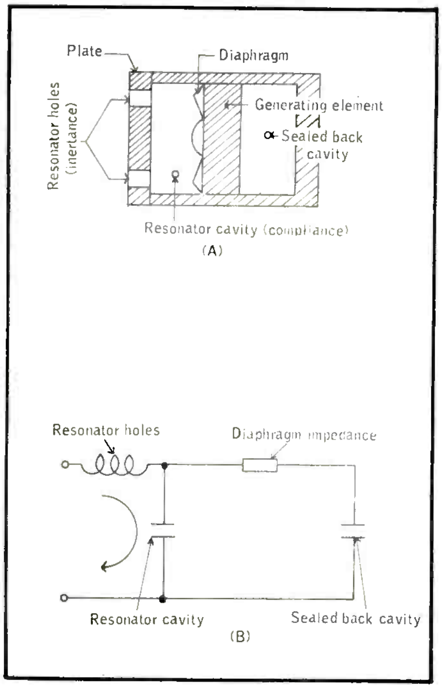


Fig. 5. The Helmholz resonator—(A) mechanical form, and (B) its analagous electrical circuit.

The desired low-frequency range then determines the volume of the microphone case, thus the only dimension not discussed—the length of the microphone case—has been determined. The over-all length of the microphone is 5 15/16 in.

When the 635A is not being handheld, it can be stand-mounted in either the Model 310 clamp, as shown in *Fig.* 1, or in the snap-out Model 311 shown in *Fig.* 4. Both of these clamps have a ⅝-27 coupler, and both have a swivel action which permits tilting the microphone to any desired angle. The 311 clamp allows the performer to snap the microphone in and out easily. The total weight of the microphone is only 6 ounces, which is not tiring to hold for an extended period of time, even in the female hand.

To help reduce weight and manufacturing cost, no transformer is used in the microphone case. #48 copper wire, (.0013 in. diameter) is used in winding the voice coil. The impedance of the voice coil is 150 ohms, which is the terminating impedance of the microphone. Eliminating the transformer also eliminates a possible pick-up of stray magnetic fields.

There is only one opening to the generating element, which makes the 635A an omnidirectional microphone. At the high-frequency end of the audio range the polar response becomes slightly directional. This is due to the baffle effect of the front of the microphone case. There is a controlled, very slight leak which allows for pressure equalization on both sides of the diaphragm. This is required as the atmospheric pressure varies to avoid any static pressure on the diaphragm.

High-frequency response is limited by the mass of the moving system—the major mass being the voice-coil. Aluminum wire was eliminated because a 150-ohm coil was desired. This requires aluminum wire finer than practical to use in production. Copper wire has greater conductivity than aluminum so that for equivalent voice-coil impedance and number of turns, a finer wire than the equivalent aluminum would have to be used. Copper is much more ductile, and smaller sizes are easier to work with.

Additional support is given to the response from 8000 Hz out by the use of a Helmholtz resonator. This resonator fits into the equivalent circuit of the microphone acoustic system as shown in *Fig.* 5.

The Helmholtz resonator is formed physically in the front of the microphone as shown at (A) in *Fig.* 5. The inertance of the holes (equivalent to inductance), resonates with the compliance of the air volume (equivalent to capacitance), between the retainer and diaphragm, as shown at (B).

Response

The response of the total microphone is shown in *Fig.* 6. As shown, the response is slightly rising and very smooth. This response was selected after many field tests because the slight rise from 2000 Hz gives increased presence and articulation without a harsh sound. The low-frequency response is shaped to control rumble without unbalancing the useful low-frequency response. Without the Helmholtz resonator and with a reduced back cavity, the response would be as shown dotted.

The smoothness of the response curve is important because the polar response is omnidirectional. In field tests the 635A has been found superior to some directional microphones on feedback; the latter often have a peaked frequency response in a band where the polar response is also weak.

Production

In production of the 635A, the generating element is thoroughly tested before assembly in the final microphone. Damping of the diaphragm is accomplished by inserting a felt material between the diaphragm and back-case cavity. This felt is then adjusted acoustically with the particular diaphragm coil that has been installed in the generating element. At this testing period the response of the generating element only is checked for smoothness of response and output sensitivity. It is then placed in a 130-dB sound field, and if any distortion is shown, the element is rejected.

If the microphone has passed all of the above tests, it is then installed in a final microphone case, placed in an anechoic sound room and checked for frequency response in a free sound field.

Periodically, a 635A is sampled from the final test and given a drop test. The drop test consists of ten drops at random angles from a height of four feet onto a hardwood floor. The sampled microphone is then given the final frequency-response test again. A visual inspection after dropping, gives indication of inferior case parts or materials.

A 635A is a microphone in action. It is equipped with a durable cable which will withstand much use.

The polarity of the microphone from the activating acoustical signal on the generating element to the end of the cable is maintained. This allows multiple microphone users to have all microphones in phase with each other.

The increased frequency with which the 635A is seen in everyday activities is the final test of it's ability to serve as a new and useful audio tool. Æ

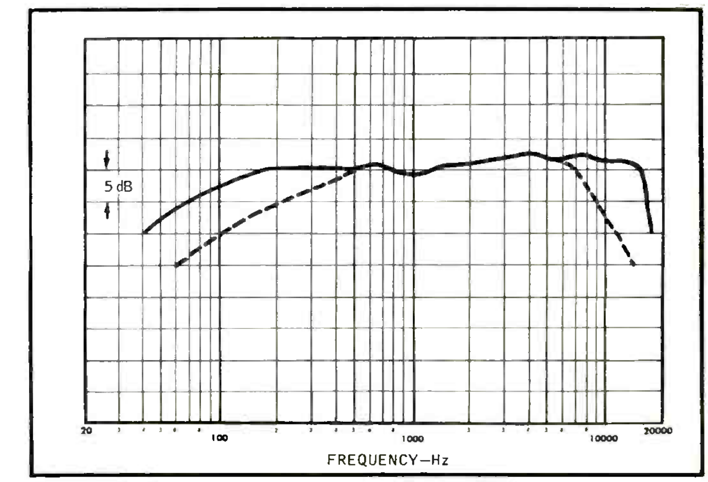
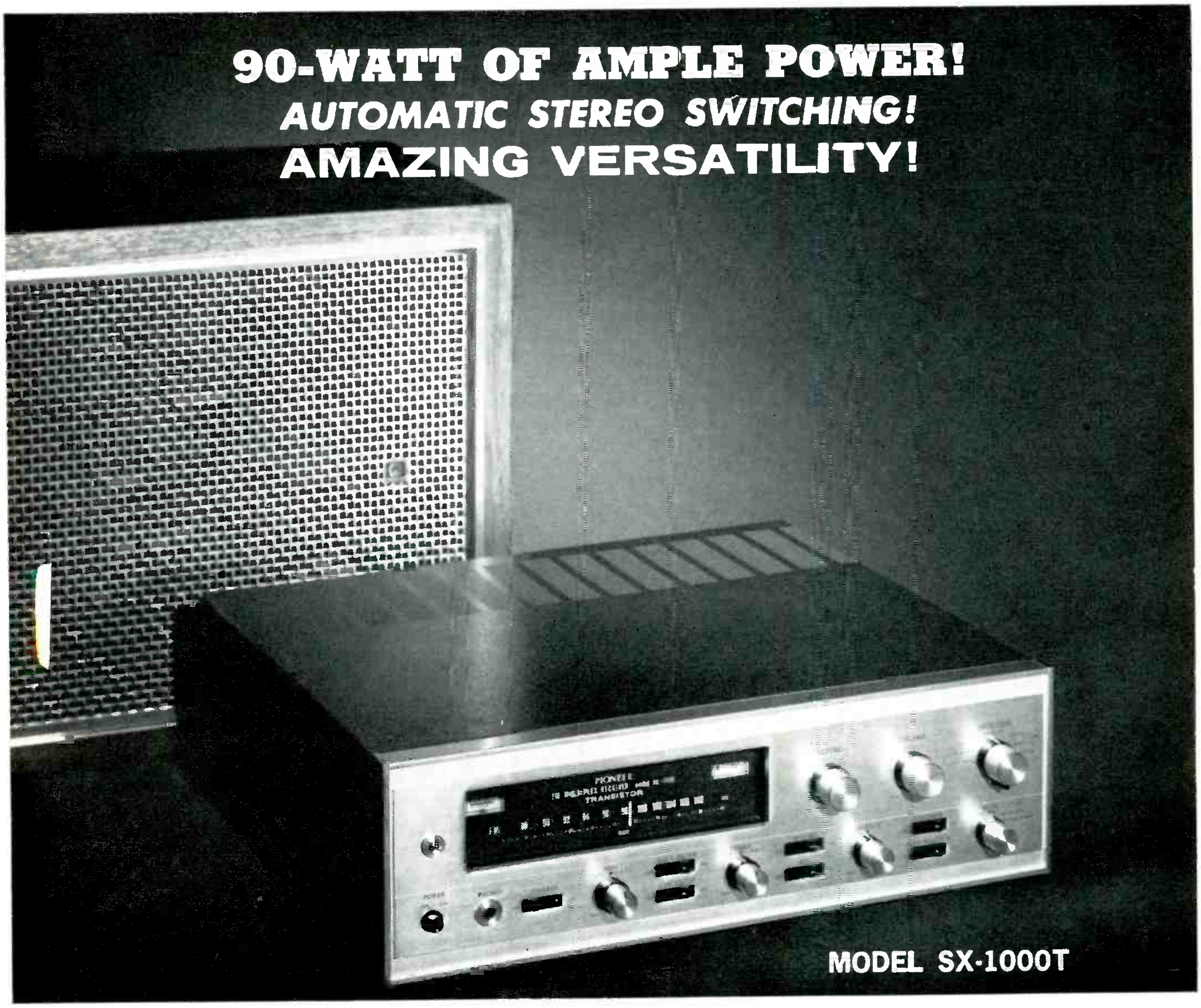


Fig. 6. Frequency response (solid line) of the 635A, compared to the response (dotted line) without the resonating back cavity.



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The above-mentioned are but a few of the superb features of PIONEER'S handsome new stereo receiver, the SX-1000T, a feature-packed receiver for the discerning listener who wants true professional performance at a practical price.

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SOUND AND SIGHT

HAROLD D. WEILER

Since Thomas A. Edison was an avid reader of scientific journals it is unlikely that he missed Obelin Smith's article on magnetic recording which we reprinted in part last month. It is interesting to conjecture what would have been the result if Edison had applied his tremendous knowledge of magnetism and electricity to this form of recording instead of his acoustic recorder. Would his motion picture camera (illustrated below) have resembled the recently released Norelco video camera? He also, of course, would have had to invent the vacuum tube.

To return to the Edison kinetoscope, even before Caveat 110 was completed he began to improve and perfect the instrument, for its closing sentence reads, "By using very large transparent shells (of glass) the pictures may even be projected on the screen as in micro-photographic projection or enlargement, the cylinder being revolved and the source of light inside the cylinder, negative records only being recorded."

While each of the previous experiments had resulted in improvements there were still a number of problems plaguing Edison, despite his optimism: the lack of concentricity of the plaster cylinders, their comparatively rough surface and the still remaining problem of focusing the microscope on the curved image.

Before Edison had an opportunity of employing glass cylinders a paper was read before a group of photographic enthusiasts at the Franklin Institute in Philadelphia on the evening of November 21, 1888, by John Carbutt of the same city. It described his new invention "The Perfect Substitute for Glass" (photographic plates). It consisted of a coating of photographic emulsion on thin (0.01 in.) flexible plates of nitrocellulose. The paper was published the following month in the *Journal of the Franklin Institute.* It was at this point where Edison's reading of scientific journals served him well, for he evidently read the paper, and quickly ordered the new material. It provided a perfect solution to his immediate problem.

A large cylinder similar to that illustrated last month was slit to accommodate Carbutt's new sensitized celluloid film. Images were recorded, the film developed and placed on a newly constructed transparent drum. This time the source of illumination, the Geissler tube, was inside the drum. The pictures were again superior to those obtained previously. The results were slow, but encouraging. Each new experiment resulted in another improvement. Edison, however, was evidently still dissatisfied, for the experimentation continued into the new year of 1889.

The reason for Edison's dissatisfaction becomes obvious when we read Caveat 114 which was sent to his patent attorney on February 3, 1889, conceived in January, 1889. It describes a new type of cylinder. The old problem of proper focusing of the microscope on the round cylinder was evidently still annoyingly present for Edison wrote, "This gives a flat face for the photographic record from the microscope and the picture is not thrown out of focus as it would be if the cylindrical surface were round, especially on very small cylinders which it is necessary to use on commercial apparatus."

This experiment was evidently a failure, for according to Edison's Caveat 114 dated May 20, 1889, he had returned to the transparent cylinder. Larger images were required but the rotundity of the surface and the resultant difficulty in obtaining proper focus precluded their use. It was exasperating for when the images were viewed flat (not mounted on the cylinder) they were perfect.

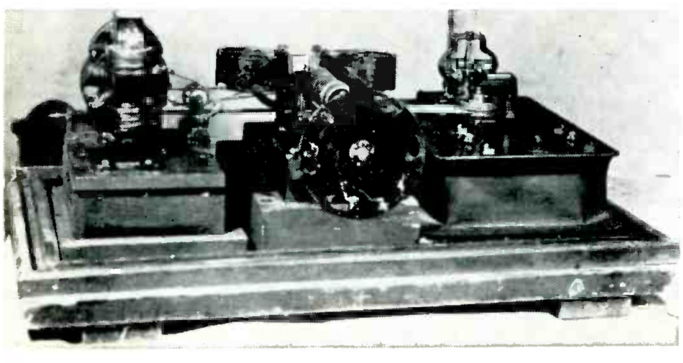
We know that the invention of the Kodak camera could not have escaped Edison. It was completely described and illustrated in *Scientific American*, September 15, 1888. Here was a device which could store enough film to take 100 2½"-diameter pictures in a space 6¾ x 3 x 3½ inches—more film than they required. So on the morning of June 3, 1889, the Eastman Dry Plate Company received a letter inquiring as to the price of the camera from the Edison Phonograph Works.

The Kodak Edison acquired was evidently obtained from Scovill Manufacturing Company some time between June 7, 1889, and the time the second one was acquired on November 23, 1894. It was a truly marvelous piece of equipment for the time, as revolutionary as Kodak's recently introduced Instamatic cameras of today. For the best description of the cameras available prior to the introduction of the #1 Kodak and its advantages over them, we must quote Ackerman's "George Eastman." Heretofore, the so-called "detective" camera has been the only instrument suitable for wayside photography and even *with this form of camera it has been necessary to perform upwards of ten operations in order to make one exposure,* as the mere act of "taking" the picture is called. The omission of any one of these consecutive operations entailed a liability of total failure of the picture."

"*The Kodak reduces the ten or more operations* heretofore necessary to make an exposure with detective cameras *to three operations* . . ." These three operations were widely advertised: "1—Pull the string. 2—Turn the key. 3—Press the button." With practice one could take three, perhaps even four pictures per minute!"

When we read this description of the state of the photographic art at the time, we cannot help but marvel at Edison's genius for here was a brand new development with which as many as three or four pictures a minute could be taken. Edison had already taken more than double this amount per second. However, we firmly believe that it was Eastman's Kodak #1 which inspired Thomas A. Edison to change from his cylinder concept and adopt a strip of film.

W. K. L. Dickson's description (*J.S. M.P.E.*, December, 1933), of the next step in the invention of the kinetograph seems to confirm this theory; he writes, "My next attempt, after abandoning drums and the like was to proceed with narrow strips of Carbutt celluloid, 18 inches long, notched on the top, and impelled intermittently by a clock escapement movement. A rotating shutter and a lens of 1½-in. focus were used. The pictures were ¼-in. square. This tentative test or experiment seemed to be leading me in the right direction, as will be shown." Slowly, laboriously, step-by-step each obstacle was overcome. Through the use of a strip the comparatively flexible film was held flat at the exact moment of exposure or viewing. Dickson's article continues, "On trying to join these strips, the usual trouble was that the joints stuck in the frame or open guide, which, however, we made as springy as possible."



Edison's motion picture camera mentioned herein.

Another problem quickly became apparent, the required stop-and-go motion quickly tore the notches from the film and the film itself. Some other means of moving the film was required. The automatic telegraph Edison had invented earlier provided him with the solution, for its paper tape was moved with a sprocket engaging perforations in the paper. This method was quickly adopted. A perforator was made and the film punched. A new sprocket was substituted. However, the problem of the joints between the short film strips remained until the Eastman Company announced in early June, 1889, their transparent film. Æ

Some plain talk from Kodak about tape:

The lowdown on low-noise tapes... and on low-speed tapes

Designing a "low noise" tape is a bit like trying to fit a six-foot man with a pair of pants tailored for a five footer. Cutting off his legs is a solution . . . but it lacks elegance. Tapewise, if all you do is use a low-noise tape, you end up with lowered output; i.e., mighty short legs. And if you push up the gain, where's the low noise you were hoping for?

The art of low noisemanship requires a bit more finesse. And it's not so hard to master if you take a listen to KODAK's Type 34A Hi Output Professional Tape. Try this test: Listen to a "no signal" tape at high gain. Now turn down the gain until the hiss disappears. Wouldn't it be nice if you could listen to the tape that way? The solution, obviously, is to pick a tape you can put a lot on—and play it back at low gain . . . and low noise, naturally!

Enters the star. Compared to our own Type 31A Standard Play Tape, and to the low-noise product from a competitor we must keep mum about, the chart below reveals that KODAK Type 34A Hi Output Tape gives five or more additional decibels of undistorted output. At similar output levels, Type 34A is just as quiet as the next fellow's. It does this with no increase in print-through over general-purpose tapes. Pretty nice for silence lovers. The values expressed in the chart are in decibels at optimum bias settings using our Type 31A as the reference.

Some like it slow. In medieval times, a favorite subject of theological discussion was just how many angels could dance on the head of a pin. KODAK can provide no informed opinion on this question, but leaps into the fray when it comes to how much signal you can squeeze on a given length of tape. Since tape started, tape speeds have been dropping. First it was 15 ips, then 7½ ips; the day of 3¾ ips is here for some. And the recorder manufacturers still haven't stopped. Who knows where it will end.

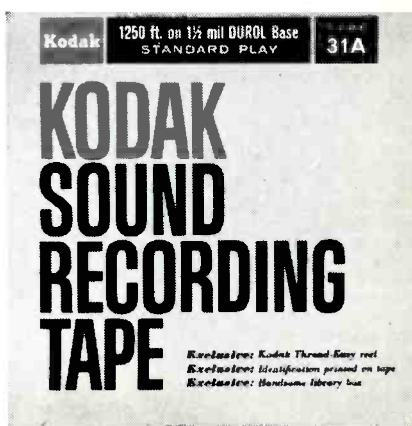
But there are some problems involved. At 15 ips a single cycle of signal at 1,000 cycles-per-second covers 15 thousandths of an inch longitudinally on the tape as it travels by. At 1⅞ ips (to go to extremes) it's down to less than 2 thousandths of an inch. As a result, as tape travel speeds decrease, tape "resolution," to borrow a photographic word, becomes more and more important. A second problem is that external magnetic flux on the tape available to thread the reproduce head also decreases in proportion. This means that you need a high-efficiency tape. Last but not least, the tape itself has to be thin for maximum footage on a given reel. People buy long-playing tapes because they play long.

Put all these problems together and our trusty KODAK 11P ½ Mil Double Play Tape sounds better and better. Look at the chart which compares it to a premium-priced famous name brand recently improved for low speed . . . and to a competitive general-purpose tape. KODAK 11P shows off as well as the first, and better than the second. Figures are in decibels using our 11P as the reference.

	Competitive double-play tape	Premium-priced competitive "improved" low-speed tape	KODAK 11P double-play tape
Optimum bias	+0.5	−0.5	0.0
Sensitivity at 37.5 mil wavelength	−0.6	−1.2	0.0
1 mil wavelength	−2.5	−0.2	0.0
0.6 mil wavelength	−2.6	+0.4	0.0

KODAK Sound Recording Tapes are available at most electronic, camera, and department stores. New, 24-page, comprehensive "Plain Talk" booklet covers all the important aspects of tape performance, and is free on request. Write: Department 8, Eastman Kodak Company, Rochester, N. Y. 14650.

	KODAK 31A Tape	Premium-priced competitive low-noise tape	KODAK 34A Tape
Bias	0.0	+0.4	+0.8
Sensitivity at 37.5 mil wavelength	0.0	−3.0	+2.1
Input at 2% harmonic distortion	+10.0	+11.4	+13.0
Output at 2% harmonic distortion	+11.5	+10.7	+16.3
Saturation Output	+20.0	+19.0	+23.6
Maximum Dynamic Range	75.0	79.0	79.0
Modulation S/N Ratio			
—20 to 1000 CPS	62.0	57.0	62.0
—1000 to 15,000 CPS	64.0	65.0	67.0



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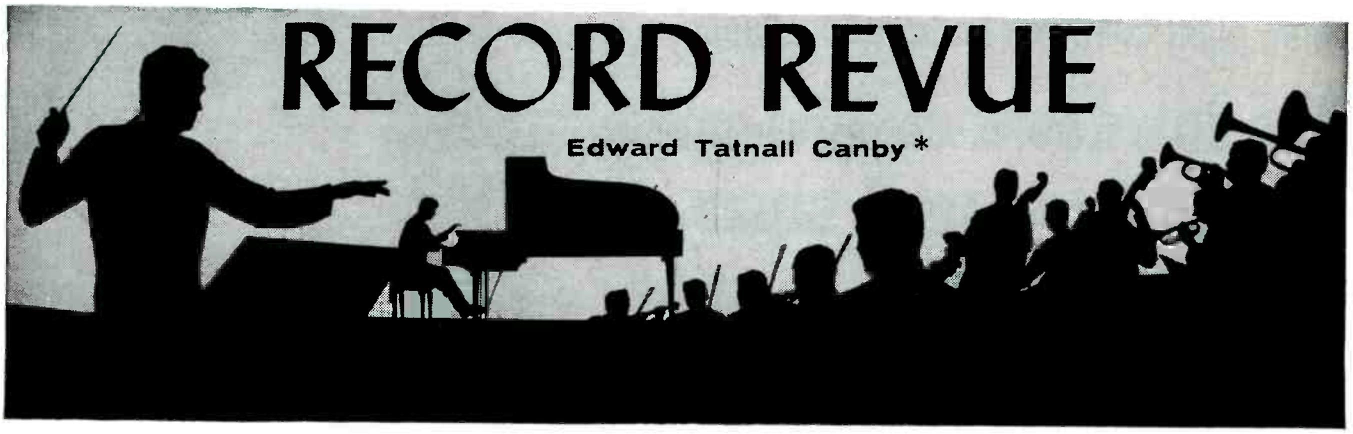


FROM BRITTEN TO DENMARK

Britten: Rejoice in the Lamb; Missa Brevis; A Ceremony of Carols. Choir of St. John's College, Cambridge, G. Guest.

Argo ZRG 5440 stereo

Britten is the outstanding British composer of the last quarter century or so—and his music is British beyond compare. So unlike anything we can turn out over here! It is modern, and yet traditional; it can be difficult but it is always for *people*, both to perform and to hear, and it is above all real vocal music, in the great British tradition. We have no such thing to fall back on, ourselves, except in our rich areas of folk music.

What a splendid record! This is a vibrant, vivid choir, the usual brace of little boys plus a few male countertenors and the lower parts, tenors and bass. But what strong singing! These kids simply light into the music, and so do their elders. As you listen, here, it is wonderfully evident that, whatever we may think of Britten as a "classical" modernist, he hits the musical nail on the head in his homeland. The music suits the performers, they blossom out for it, they understand every note.

Quite a variety. The familiar *Ceremony of Carols* has a rousing performance, almost raw in its fervor. No sweety-pie little angel-boys here. The *Missa Brevis*, a short Mass, is for three-part boys' choir with supporting instrumental music. The Cantata *Rejoice in the Lamb*, an earlier work (1943), goes back to a late-Victorian sort of writing, with much rousing solo music—it's a setting of the delightful 18th-century religious poetry of Christopher Smart, who was smart but an inmate of an asylum; its most famous passage is the soliloquy on his Cat Jeoffry and its natural enemy, the Mouse. Charming music, cat-like and mousey. Altogether a superb choral recording.

Britten: Part Songs. Elizabethan Singers. Halsey.

Argo RG 424 mono (also stereo)

Here is the British composer at home in his most versatile medium, adult choral music. These assorted choruses, unaccompanied and with piano, are virtuoso affairs, showing off every facet of choral expression. *Five Flower Songs*, to ancient English words, are relatively serious works of contrapuntal complexity—except the last, a whirlwind thing about Green Broom. Then comes a group of songs with piano (Robert Graves, W. H. Auden) and, on Side 2, a very un-folk-like setting of the *Ballad of Little Musgrave* (often heard in a U.S. folk version) and a group of rather glutinously rich "choral dances" out of the opera, *Gloriana*, composed for Elizabeth's 1953 coronation. Finally, a pair of stunt folk-type pieces (*not* American-type folk but traditional English!), ending with *Oliver Cromwell*, in which is featured the fastest sung English I've ever heard. *OliverCromwelllayburiedanddead. Theregrewanoldappletreeoverhishead . . .*

Singing by the medium-large chorus is very expert in both the slow and rapid music but a bit unblended, with solo-type voices that tend to stand out, plus a good deal of compound vibrato. Doesn't matter—the group sings well and intelligibly, and with excellent humor.

The Brontës. A dramatic reading by Margaret Webster.

Vanguard VRS 9176/77 (2) mono

Recordings of drama are coming in thick and fast—they seem to be the latest sensation, and indeed they do offer terrific entertainment value. The Brontë family was that obscure, ingrown houseful of strange English creatures, in the early 1800's, a parson father, a whole brace of girls and one boy, who out of almost nothing turned into the century's most brilliant novelist family—Charlotte, Anne, Emily—capped by their most famous book, "Wuthering Heights," The girls wrote delicately under male pseudonyms, for Victorian ladies weren't supposed to write novels.

They lived in utter isolation in a tiny village, except for brief spells as governess to spoiled brats, or at schools, dreadful places; and, one after the other, they died, dreadfully, languishing away. It's all in this really absorbing spoken tale, in a kind of journal form, out of letters, biography, writings, put together for "live" stage reading by Margaret Webster and done here—just the one, single voice—to perfection. It's an evening's wonder, and will leave you thinking for many a day after.

Bernstein Nielsen Symphony No. 3. Royal Danish Orchestra. **(With booklet)**

Columbia MS 6769 stereo.

Nielsen (1865-1931) was Denmark's only composer, at least for the outside world; and when Bernstein conducted this symphony (1911) recently with the Danes' own special orchestra—it practically broke up the country, they were so excited.

Well, I love the Danes but I dunno. He was good, all right, but he's awful dated right now, Nielsen, unless you like this kind of thing. A real, old-fashioned late-to-neo-Romantic. Lots of people are swooning about the music so you'd better take it seriously, especially if you like, say, Sibelius and Rachmaninoff. Nice hidden (wordless) soprano and tenor in the second movement.

It *is* a stirring performance. You can just hear the faithful Danes rising to the great occasion. They never heard it so good.

Nielsen: Symphony No. 4 ("The Inextinguishable"). Hallè Orch., Sir John Barbirolli.

Vanguard Everyman SRV 179SD stereo

Well, I haven't extinguished this one yet—I'm listening as I write. It's from 1916 and reflects, they say, some of the emotions of World War I. To me it sounds a bit more "old-fashioned" than No. 3, even more Romantic but, in its Romantic way, quite lovely. Also quite long, full of typical late-Romantic wanderings all over the lot, full of tuneful ideas, oddly like Shostakovitch at times but more mellow, more human. Very moody, very passionate—*better* than Shostakovitch (for my ear) . . . but again, I dunno. You have to like big-scale, long-drawn-out symphonic Romanticism to fall head over heels for this excellent Dane. I do enjoy him here, I'll admit. More than in the Symphony No. 3 preceding.

Performance is also sincere and musically first-rate; for Sir John B. (he once also conducted the New York Philharmonic—remember?) is just of the right generation for this music. He really makes it boil and roar and sigh and weep. Excellent.

VIRTUOSO PIANO

Lorin Hollander. Moussorgsky: Pictures at an Exhibition; Rachmaninoff: Prelude in C-Sharp Minor; Prokofieff: Toccata.

RCA Victor LSC 2823 stereo

Only a few years ago, the 14-year-old Lorin Hollander sallied forth on records to do battle with the biggest pianos around, which very nearly collapsed under the attack. What a juvenile fiend! Now he's somewhat older, still smiling boyishly, neatly dressed and barbered (see jacket-cover photos) and the powerhouse attack is still there. But it is tempered—if you keep the volume down a bit—by a more mature intelligence.

Moussorsky's battle horse, originally for piano, can take the Hollander approach very nicely and does—this performance contrasts interestingly with Horowitz' famous souped-up piano version, somewhat re-transcribed from the Ravel orchestration. Plenty of drama here, but a good deal of crisp subtlety and tension too. I like it. As for the small-type "encores," the famous old C-sharp minor prelude gets an astonishingly solemn and impressive setting-forth, the well-known holy terror (for pianists) of the Prokofieff Toccata is as nothing under the Hollander steel-spring fingers.

RCA beats anybody in recording a big-piano sound. Don't know what they do but it reminds you of the M-G-M lion (when he was young and virile). Terrific.

Bach: Goldberg Variations. Peter Serkin, piano.

RCA Victor LSC 2851 stereo.

Peter Serkin is a chip off a notable block. His father is the pianist and all-around musician Rudolf Serkin, and Peter was 17 when this record was made. It is astonishingly mature and well thought out as a performance, and brilliant in technical execution, too. There aren't a dozen pianists in the country who could hope to match this sort of music-making.

The comparison will be made with Glenn Gould, who recorded a "Goldberg" as one of his first piano sensations. He, too, was a brilliant young newcomer. But Gould was (and is) far more eccentric, more self-instructed, much less aware of outside traditions and musical culture. Gould makes up his own Bach, and Beethoven, and it is musically remarkable, if outrageously un-stylish. Serkin knows better. He has heard this music and plenty more from the best sources through most of his life. So—no fancy long-hair eccentricities, though there's hair enough on Peter's head. (In fact, he looks a bit like Gould.)

Not only are the individual variations done with impeccable understanding and communication but, you will find, the whole huge architecture of this big piece is conceived as one, growing in drama and cumulative tension over two long LP sides until the final few variations (taken without a break) are breathtaking in effect. Some of the biggest players on both piano and harpsichord have utterly failed to see this dramatic whole—though old Landowska knew all about it.

Yes—the dramatic approach smacks of the Romantic. But, first, the piano is a Romantic instrument by rights and this is good pianism. Why play piano like harpsichord? Second, the drama, without the slightest question, *is* there, in the cumulative effect of these dozens of variations, these patterns-within-patterns. It couldn't be otherwise. What matters, simply, is the performer's *awareness* of this—and how he chooses to bring that awareness forward. On the harpsichord the means may be quite different than those best for the piano.

More power to Peter!

Schumann: Symphonic Etudes, Op. 13; Toccata in C, Op. 7. Gyorgy Cziffra, piano.

Decca DL 710115 stereo

Schumann's big piano music is just about the toughest music to project that there is, today. Most pianists can play the notes easily enough—but to project the spirit—fiery, yet related, tender, triumphant, honest, naïve and often redundant to dullness, takes colossal understanding and piano drama, as well as a superb ear—for Schumann's harmonies are not simple.

This Cziffra puts on a tremendous show, technically brilliant. But the music doesn't come off. Another pianistic head rolls in the dust—there have been plenty before.

Here, all is fiery Hungarian bravura, terribly intense and not at all relaxed (as I hear it). The great climaxes are furious, instead of grand, the romance is pushed too hard, the drama is unstable rather than persuasive. It goes unstable only too easily, alas. For Schumann, one must have an utterly calm, reassuring confidence, a radiantly serene personality, to balance out the high-charge, pathos-bathos music. It needs that kind of emotional support. No wonder that there have been many great woman Schumann players—like Dame Myra Hess, or, of course, Clara Schumann, the composer's pianist wife. They had it.

Nor does Cziffra do his pedaling well, and his harmony-hearing. There are microscopic chord-blurs, where each harmony should sound of itself, cleanly. And the climactic harmony-changes—that sudden astounding shift just at the end of the long *Etudes* — these are passed over as though nothing special had happened. Nothing has, if you don't hear it!

Nothing wrong with the Cziffra fingers and wrists and arms. They encompass the strenuous music with the greatest of security.

Schubert: Chamber Works for Piano Duo. Caroline Wood, Eleanor Hancock.

Collectors Guild CGC 641 mono
(507 Fifth Av. N.Y. 10017)

Two girl pianists graduate from Juillard; one of them marries a recording engineer, and pianist, too, and next thing you know—Eleanor Hancock plays, this, with Caroline Norwood; David Hancock makes the recording—and writes the musical annotations. Some family deal!

The music is for two pianists at one piano, sitting side by side. That doesn't make for a good show in the concert hall, but it is marvelous for the home musician, as thousands of people knew before the lazy phonograph came along. These girls gave a Town Hall recital, nevertheless, for which they should get medals. Most pianists are much too vain. They want the whole instrument for themselves, natch.

Schubert is tough music to play—one piano or two. Only the very greatest musical minds really bring it off superbly, both the notes and the sense. And so these two girls do really nobly. Their coordination is perfect, not a note out of place or stumbled. Their scherzi trip along neatly, their slow melodies sing out, their Schubert lightning-harmonies, those marvelously subtle changes, are always in place. If I keep thinking of the two top Schubert men in Austria, the fabulous team of Demus and Badura-Skoda, who once recorded so much for Westminster, it is merely with nostalgia and the thought that (as the ads say) only in Austria do they play more Schubert better. Nobody could beat *them*. Not even these girls.

The music includes the big B flat Sonata Op. 30, the Fantasia in F Minor, Op. 103, an Andantino Varie, Op. 84, No. 1, and a March, Op. 121, No. 2.

OLD MUSIC FROM DOVER

Guillaume Dufay: Secular and Sacred Music for Voices and Instruments. Ambrosian Singers & Players, Denis Stevens.

Dover HCR 5261 mono

Missa Salve; Missa de Sancta Maria (Anon. 13th c. Masses, England, Spain). Ambrosian Singers, Stevens.

Dover HCR 5263 mono

Secular Vocal Music of the Renaissance from Spain, Italy and France. Ambrosian Singers, Stevens.

Dover HCR 5262 mono

The low-priced Dover line ($2 list, mono and stereo) now includes a number of discs of these English "old music" performers, led by a leading musicologist-scholar who—amazing to say—is also a sensitive performing conductor. For anyone even mildly interested in this really old music, these performances are wonderfully communicative, though some are more so than others. The Ambrosian singers, solo voices, share a musical sensitivity and earnestness with their conductor, as well as first-rate musicianship in respect to phrasing and shaping. Dover includes all texts, with translations, for the low-low price.

However, there is one qualification here. Today's solo voices are trained *not* to blend together in ensemble. They invariably use vibrato, sing forcefully (to carry in the concert hall or opera house) and with an erratic and often inaccurate "line," as compared with musical instruments. It is not easy to live down this background, even with the best of intentions.

And so the finest music here is that which is sung by solo voices, not more than two at a time, with other parts done instrumentally. That includes most of the material on these discs. As soon as three or four or more voices sing together there is that unpleasant wobbly non-blend that is typical of today's professional vocalism and was clearly *not* the sound of the original music.

The early 15th century music of Dufay is superb in the first of these records above. It is a strange musical "speech" at first—but in these gentle, plastic, responsive performances it soon gets through. The secular works are done intimately close, the Latin-text pieces in church-type acoustics.

The two 13th-century Masses are astonishingly potent as well as "primitive"—primitive only in that the concept of harmony and of independent voice-lines is just beginning to show. Other than that, the music is sheer Gothic, that combination of stark, raw expression — gargoyle-like — with utter sophistication, that so impresses us in the great cathedral architecture.

The most familiar style, the 16th century Renaissance music (HCR 5262), is the least effective. Too many four- and five-part vocal ensembles, done with much heartfelt inaccuracy of detail and lots of expressive wobble, in a somewhat old-fashioned "Romantic" manner. But here, again, the partly-instrumental numbers nicely save the day. They are excellent.

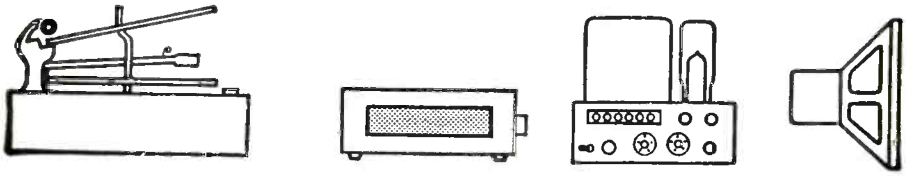
Anthology of Renaissance Music. (Dufay, Des Prez, Morley, Victoria, Palestrina, Jaquet of Mantua Mouton) Primavera Singers of the N.Y. Pro Musica, Greenberg.

Dover HCR 5248 mono

How utterly unlike the British Ambrosian Singers! These are solo "pro"

(Continued on page 65)

EQUIPMENT



PROFILE

UTC SOUND MAXIMUS 7 SPEAKER SYSTEM

This is the largest and most costly of the Maximus systems. These systems are all advertised as having the ability to get a maximum of sound out of a minimum of enclosure. As the largest of the species, the Maximus 7 is pretty much the same size as other popular "bookshelf" speakers. (*Encyclopædia Britannica* books, that is, and then some.)

By way of features, the Maximus 7 utilizes a long-throw 12-inch woofer that is pneumatically (acoustically?) suspended within the sealed cabinet. Crossover at 1800 Hz is to a pair of mid/high cone drivers with directive lenses in front. A second crossover at 8000 Hz goes to a dome super-tweeter, again featuring a pattern shaping front-piece.

The enclosure itself is of the customary rectangular shape, and is finished on all sides, including the back panel. The front grille frame is attached by snap-fasteners; it is thus readily removable.

Once off, the grille-less box reveals the four drivers and a pair of otherwise unmentioned control pots. These allow a fair degree of mid-range and high-end adjustment to suit the acoustical environment in which the speaker finds itself. Also, the bare-faced box is not unattractive of itself. The drivers are all well-finished and the wood of the front panel is the same finished oiled walnut as the rest of the enclosure. All in all, the Maximus 7 dressed or naked is complementary to most decorating schemes.

Connection is to the 8-ohm tap of your amplifier. The rear posts of the speaker are coded for phasing purposes by a red patch on one of the connection posts. This could come off, so it might be a good idea to scratch a small *x* into the woodwork under this post.

Efficiency is low. Ten good watts would be minimum, twenty would be a better idea. Weight might also be a factor if you want to wall-mount the units. They are just under 60 lbs. each. This can prove a considerable strain on modern walls unless special precautions are taken.

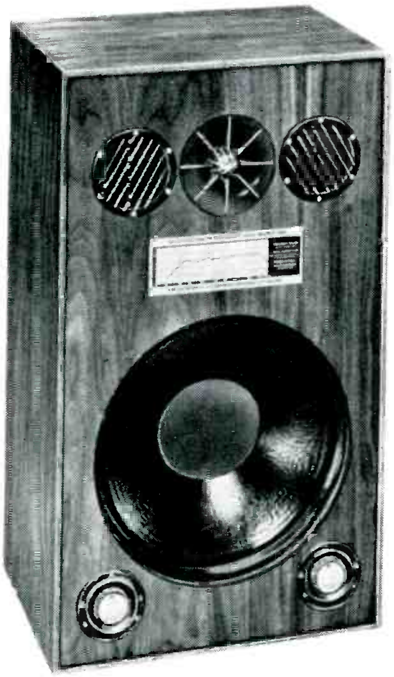
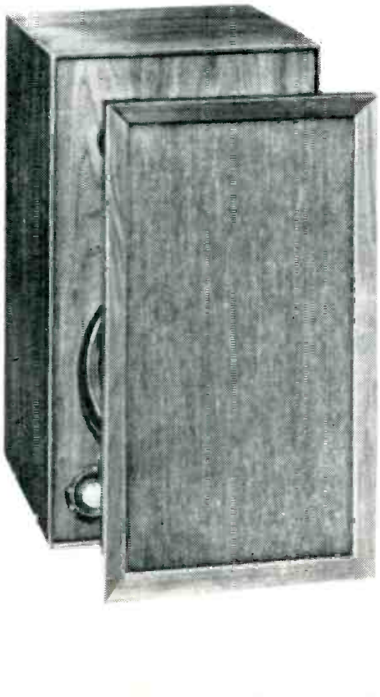


Fig. 1. Maximus 7 Loudspeaker System, without (above) and with (below) the snap-out frame and grille.



The Sound

The Maximus 7, just as its smaller relatives, is a distinguished speaker. It sounds with the authority of quality of manufacture and design. That is to say, that the music that poured forth from the pair of units we had as samples was listenable indeed.

Perhaps the most distinguishing quality is crispness. There is none of the muddiness that is so common to compact systems. From one end of the frequency spectrum to the other, sound was always clear.

Another quality we liked was the *lack* of special emphasis that could be placed (through use of the speaker controls) at any section of the speaker's range. There is that kind of evenness of response usually only associated with the finer electrostatics.

Add to this a highly respectable bass end—virtually flat to 45 Hz with a rapid roll-off below; still usable output at 25 Hz. And a top end that goes well beyond the reliability of our ears and microphones.

The sum of all of these virtues comes out a most reasonable system. Music has that open, easy quality that we have come to expect from finer systems. Perhaps the musical mid-range is a trifle rougher than our tastes ultimately prefer, but this is *taste* and not a clear-cut objective statement. In any case, the other good qualities more than overcome this feeling. We can safely say that we could live with this system. And that says a lot.

Finally, add the list price of $189.00. Total the sum of all of these parts and you will find, as we have found, that the Maximus 7 is a most worthy product, deserving of consideration from any point of view. Circle 200

KNIGHT-KIT "SUPERBA" KG-415 DELUXE STEREO TAPE RECORDER

This is a kit which is a perfect delight to profile for two reasons—it was a pleasure to construct it, and it performed so well after it was completed. We feel as though we can say this with authority, because, due to circumstances beyond our control, we actually constructed two of them within the space of one month, and both fulfilled their expectations completely.

The KG-415 consists of two separate units—the electronics package, and a Viking 87-RMQ transport, which is a two-speed deck with three heads. It employs two motors, both of the induction type, with one used for the capstan drive (and rewind), and the other for take-up (and fast forward). The three heads—erase, record, and play—are all of the "premium" type, made by Nortronics, and having the hyperbolic contour which eliminates the need for pressure pads. Two separate knobs control the tape motion—the inner one for forward record and play movement, and the outer one for the fast spooling operation. In addition, a "cue" position of the inner knob permits hand movement of the reels for editing purposes. A button in the center of this concentric knob actuates the bias/erase oscillator and simultaneously lights an indicator lamp on the amplifier panel. A three-dial pushbutton-reset digital counter is also mounted on the panel.

The Amplifier

The electronics portion of the recorder is entirely solid-state, using 17 transistors and six diodes. The over-all arrangement is somewhat unusual in that there is one rotary switch at the center of the panel, which selects the mode of operation. This switch has six positions—play left, play right, play stereo, record left, record right, and record stereo. Six little windows along the panel indicate the selected modes, one of them lighting up for each position of the switch. The play windows are blue while the record windows are red. Even though a switch is in one of the record positions, the unit does not record unless the button on the deck is pressed, and the recording action is signalled by the light at the center, directly above the selector switch.

Four other switches are mounted on the panel, as well as eight controls. The switches, of the slide type, turn on the a.c. power, select equalization for either 7½ ips or 3¾, select monitoring from source or from the tape, and set up the internal circuitry for normal operation, echo (in which the playback signal is fed back into the same channel and re-recorded), or sound-on-sound, (in which the playback signal from the left channel is fed back to the recording circuits of the right channel, and vice versa).

The two small knobs adjacent to the level-indicating meters permit the adjustment of the source monitoring level to equal that of the playback level. The knobs along the bottom are, from left to right, left playback level, left line recording level, left microphone recording level, right microphone, right line, and right playback. The three jacks serve for microphone inputs, left and right, with the center one for headphone monitoring.

The rear apron of the chassis mounts a pair of line inputs, a pair of output phono jacks, a pair of monitor phono jacks, a test-oscillator switch, a test-oscillator output jack, a pair of convenience outlets, a fuse, and the line-cord. On the top of the chassis are three pairs of phono jacks for the three heads, and two single jacks, one for the bias/erase oscillator control cable from the deck and one for the record indicator light cable.

The test oscillator is a transistorized circuit which provides a test signal of 1000 Hz at slightly over 0.5 volts for various alignment operations. This is the only transistor circuit actually constructed on the chassis. All the rest are built in module form on six separate printed-circuit boards which plug into connectors on the chassis. Four of these are identical, each employing three transistors, and serve as record and playback amplifiers. Differing types of feedback are applied around these circuits for play and record use, but this feedback is applied at the connectors so the modules remain interchangeable. The high gain required for playback is also necessary for the microphone for recording.

A fifth module is the stereo headphone amplifier, which provides a low-impedance output for use with conventional 4- or 8-ohm phones. The sixth module is the bias/erase oscillator, using two npn silicon transistors in a push-pull circuit operating at around 80 kHz. All other transistors are pnp germaniums. Two additional small modules provide the required equalization for recording.

Fig. 2. Knight KG-415 "Superba" Solid-State Tape Deck Kit.

Construction

Over-all construction time was just over 20 hours for the first set and about an hour less on the second. Instructions are simple and clear, and exceptionally accurate. Knight-kits generally provide pre-cut and pre-stripped wire, which makes it possible for the instructions to specify, for example, "Green wire. Solder one end to terminal 8 of S-4. Route the other end as shown; it will be connected later." Most wire lengths follow the EIA coding —red is 2 in., orange 3 in., and so on. Where there are exceptions, the length is given. Brown is 10 in., white with brown tracer is 11 in., continuing as far as necessary. The wire furnished is a new type which we have not heretofore encountered in kits—it seems to be lightly soldered, so that although stranded, it is nearly as stiff as solid hookup wire, and thus makes for neat construction because the wire stays where you put it. Another advantage is that it does not fray out as you attach it to a terminal. The shielded cables supplied are also cut to length and stripped, so that the drudgery of wire measuring, cutting, and stripping is entirely eliminated.

The bias and erase currents are individually adjustable for the two channels on another module, which like all the amplifier modules, is protected by a metal shield.

Side brackets attach to the chassis and support the deck above it, and the whole recorder may be mounted in an optional walnut-finished wood base or in a portable case, also optional.

Interconnecting cables, already terminated, are furnished for making the connections between deck and chassis, and these, along with the power cord from the deck motors, are held neatly against the brackets by plastic cable clamps.

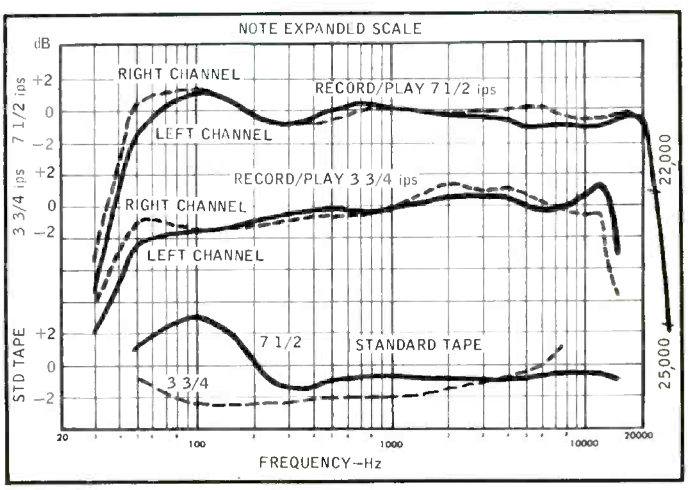


Fig. 3. Frequency-response performance of the KG-415 in various modes.

Performance

Considering the minor (?) variations in transistor parameters, it was amazing how nearly identical the individual amplifier modules measured—within 0.5 dB throughout, and all the way from 30 to 25,000 Hz. *Figure* 3 shows the frequency response of the recorder/amplifier assembly in three sections—left and right channels at 7½ ips; left and right channels at 3¾ ips; and response from Ampex Standard tapes 31321-01 and 31331-01, averaged between the channels. The bump at 100 Hz on the 7½-ips curve we have seen so often on many different recorders that we have come to believe that the standard tape has the boost, rather than all of the recorders. Note the expanded scale. If these curves had been plotted with the usual spacing of 5 dB between the horizontal lines, they would have appeared to be straight lines almost—to show how close the channels actually are over the entire range, we expanded the dB scale as indicated. This is exceptionally good performance for any non-broadcast machine, in our opinion. At an indicated record level of 0 VU, the distortion through the recording amplifier, onto the tape, and off through the playback amplifier measured 1.1 per cent, and it reached only 2 per cent at an indicated record level of +3 UV. Signal-to-noise ratio measured 52 dB referred to a 0-VU signal, when using Scotch 202 low-noise tape. Flutter and wow came to slightly under 0.2 per cent at 7½ ips. At the relatively low price of $249.95 plus some 20 hours of pleasurable work, we think the KG-415 is an excellent buy.

On our second machine—we had to part with the first one, but measurements were practically identical—we elected to install Preh 6998[1] panel-mounting sockets for microphone inputs, since all our microphones terminate in Preh plugs. The "hot" connection on these plugs, as used in other machines so equipped, is terminal 3, with terminal 2 the common or ground, and terminal 1 is normally unused. We connected a 0.12-meg ¼-watt resistor between terminals 1 and 3, thus providing another high-level input to each channel when not using the microphone. Access to this terminal can be had by using any 3-pin Hirschmann or Preh plug with the high-level circuit feeding into terminal 1. With this modification, it is possible to mix two high-level sources into each channel, although still retaining the ability to mix a microphone with a single high-level source. This strikes us as a desirable modification—though we haven't yet had occasion to use it. But we could, if we needed to.

With the faces of the heads upward, and the head cover being hinged so as to lift up when desired for cleaning, adjusting, this machine is easy to use for editing.

Mention was made of the monitor and output jacks on the rear apron. These are actually fed by the same circuitry, but the output jacks are connected only when the selector switch is in a play position. The monitor jacks are "hot" at all positions of the switch. Circle 201

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[1] Available from Switchcraft.

SONOTONE RM-1K SPEAKER SYSTEM KIT.

It isn't too often that we can say that a kit was entirely built while we were watching Roger Moore cavort as Leslie Charteris' *The Saint*. And the only reason that the kit took as long as a full hour is because every once in a while Simon Templar succeeded in distracting us completely from our duties.

This is an easy and fast kit, but we really do not suggest construction concurrent with TV. The kit really does deserve sole attention. Its low price belies the fact that the finished product is quite a decent product. But we are getting ahead of ourselves.

To begin with the kit is very well packed. All of the parts are contained within the cabinet which is already constructed and sanded smooth, ready-to-finish. The cabinet in turn, nestles behind a thick layer of protective packaging material. Before you do any thing, stop. Read the instruction manual.

On the very first page, in the very first instruction, is the instruction: "Put on gloves supplied *before* handling the Fiberglass (*sic*) acoustic material."

Good advice indeed. Glass fiber can cause nasty splinters. Sonotone has provided a pair of thin gauge plastic gloves. They are not much good for anything else, but they will see you through the kit.

Construction involves mounting the amplifier connection block, crossover coil and capacitor, and control pot on the rear panel. During installation of the amplifier block you can make later wiring easier for yourself by straightening the lugs. (As received, they will have a slight upwards bend. Left this way, the lugs contact the wall of the back panel making it difficult to insert a wire into the lug holes later on.)

The 6-inch high-compliance woofer and 1½-inch tweeter mount on the cabinet front panel. In the woofer mounting care must be taken that the screws coming up from the panel do not puncture the woofer cone. Also, we found on our samples that the woofer (which fits into a slight well in the front panel) had to be rather firmly pressed into position because of a lack of speaker, screw hole, and well alignment.

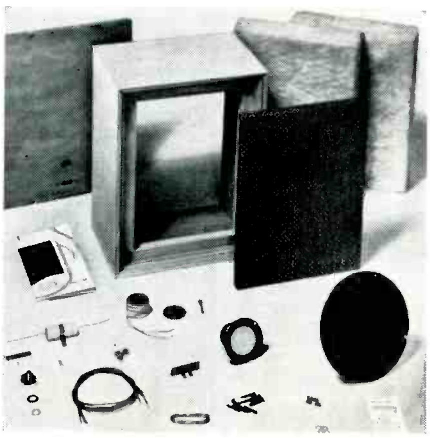


Fig. 4. Sonotone RM-1K Speaker System Kit.

Also in mounting the tweeter, which screws to the panel with wood screws, the panel holes provided and the speaker chassis holes did not quite line up. This is of no real consequence. The wood screws can easily cut new holes.

Actual wiring is simple. Most wire has been cut to length, but they have not been stripped. There are no tight spots to wire except the aforementioned amplifier block.

All that's left now is to place the two Fiberglas bats into the enclosure. *Use those gloves*! Now screw on the back panel and the job is complete.

Note that we have not mentioned the finishing of the box. That is beyond the hour's time we have discussed. The smooth, bland grain takes well to grain treatments or painting. The instruction manual does not go into great detail on this subject except to point out that you can do your finishing before or after construction. It is actually easier before the front and rear panels are assembled.

The Sound

Building is one thing. But then, you must listen to the speaker. And this is what determines the ultimate worth. It must be understood that this is a speaker system that lists for all of $35.50. That is not much to spend for a speaker. So, we expected very little when we connected them to the 8-ohm taps of our amplifier. What a pleasant surprise!

No, this is not a great speaker. It has no *deep* bass, the upper registers of music are a bit rough and somewhat peaked. But the over-all effect is good. Transients are excellent. The frequency range seems much wider than it really is; this is particularly true of the bass end. With its weaknesses, this speaker still remains easy to listen to. There is no gross unpleasantness about the sound. Rather, music, is projected forward a bit but is otherwise very smooth.

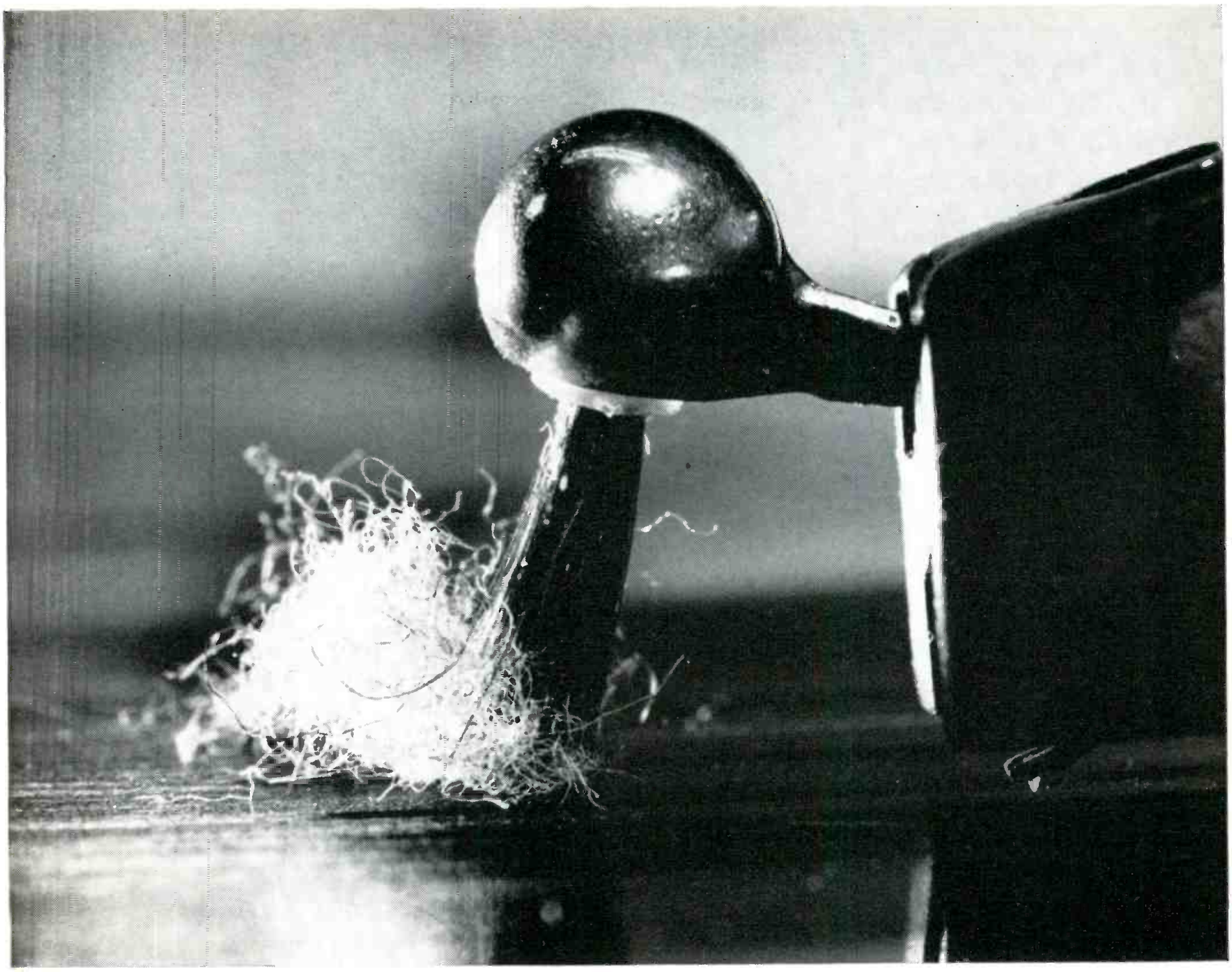
Frequency sweeps showed that the overall range is indeed as smooth as it sounds. There is a rise at 3-4000 Hz, and above that it extends well out to 15,000 Hz, albeit somewhat subdued.

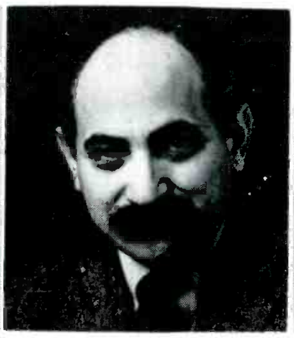
All through the musical midrange, sound is level. Going down into the bass region finds good output to 80 Hz. Falloff below that is rapid, with an effort of output still there at 40 Hz. Below that—nothing. Heavy bass pushing of the long-throw woofer showed that it had high resistance to doubling effects or other distortions. These are some of the reasons why this speaker produces the musical effect of bass in excess of what is really there.

The finished speaker is of low efficiency. It wants an amplifier capable of ten watts of music power, particularly at the bass frequencies. Given a good source, the speaker responds with good sound.

Again, it must be emphasized that this is a small and inexpensive component. There really is no right to expect much from it. So it is all the more pleasant to report that it is indeed a product that our ears found produced almost *livable* sound, which makes this kit quite a bargain. Circle 202

(*Continued on page* 63)





JAZZ and all that

Bertram Stanleigh

Benny Goodman: The Small Groups
RCA Victor Vintage Series Mono LPV521

These are reissues of the glorious quartet, quintet, and sextet waxings that Goodman made for Victor in the middle thirties before his switch to Columbia. Of the sixteen tunes on this disc, *China Boy, My Melancholy Baby, The Blues in My Flat, The Blues in Your Flat, Vieni, Vieni, Dizzy Spells, Exactly Like You, I Cried for You, All My Life,* and *Stompin' at the Savoy* have never previously appeared on LP. The performances feature Teddy Wilson, piano, and Lionel Hampton, vibes, with Gene Krupa on drums in most instances. Dave Tough is on skins for three numbers recorded in 1938, and a fourth waxing from that year has drummer Buddy Schutz. There are vocals by Hampton on *Exactly Like You, Vibraphone Blues* and *The Blues in My Flat,* Helen Ward on *All My Life,* and Martha Tilton on *Bei Mir Bist Du Schön,* which also features Ziggy Elman on trumpet. I'd expected to find these performances a trifle dated after all these years, but I was completely bowled over by the vigor on these exciting discs. Although they were all recorded between 1936 and 1938, sound is quite adequate—sometimes better than on the original shellac pressings.

Al Grey: Shades of Grey
Tangerine Mono TRC-1504

Al Grey, trombone, is joined by a group of veteran jazzmen who have in most cases worked with the Basie band at one time or another—Harry "Sweets" Edison, trumpet, Eddie "Lockjaw" Davis, tenor, Elvira Redd, tenor, Grover Mitchell, trombone, William Hughes, trombone, Kirk Stuart, piano, Wyatt Ruther, bass, Sonny Payne, drums, Rufus Jones, drums, and an unnamed organist. While everyone involved turns in the kind of polished performance that can be expected from such experienced professionals, the results are less than exhilarating. In his solos Grey has a penchant for stressing technical gymnastics for no other reason than to display his virtuosity, and the ensemble arrangements by Roger Spotts (is he the anonymous organist?) are rather thick and heavy. Sound on this mono version is on the same level as the rest of the production—smoothly professional but dull.

The Jazz Crusaders: Chile Con Soul
Pacific Jazz Stereo ST-20092

The Jazz Crusaders—Winton Felder, tenor, Wayne Henderson, trombone, Nesbert Hooper, drums, and Joe Sample, piano—are augmented by Hubert Laws, Jr., flute, Al McKibbon, bass, Hungaria Garcia, timbales and cowbell, Carlos Vidal, conga, and Clare Fischer, organ, in a sprightly group of Latin-influenced improvisations that are always fresh and original. Laws, who had worked with the Crusaders when they were the Modern Jazz Sextet, is right at home here, particularly at such moments as his flute is paired with Joe Sample's piano. No one pushes too hard, and the results are a most engaging blend of solo and ensemble. The recording is spread out, with a slightly distant quality, except for flute and percussion which are prominently forward and are reproduced with an abundance of crisp transients.

Gil Fuller and the Monterey Jazz Festival Orch. Featuring Dizzy Gillespie
Pacific Jazz Stereo PJ-93

The only real interest here is in Dizzy, even though Fuller and the band get top billing. Fuller had done arrangements for the Gillespie band of the late forties, and it was Dizzy who selected him to serve as musical director and coordinator of the 1965 Monterey Jazz Festival. His arrangements are smooth, and the band is slick, but there is a subdued quality to all of its contributions that robs this platter of the excitement it ought to generate. Dizzy's solos are another matter. He's in good shape, and his performances are as fluent and breakneck as ever, even though rather distant miking suggests a more placid quality than is characteristic of him. A new version of Dizzy's *Groovin' High* will be of immediate interest to all admirers. It's a much more mellow account than the shellac version made with Charlie Parker in the early forties.

The Rod Levitt Orchestra: Insight
RCA Victor Mono LPM-3372

About a year ago, Levitt's first recording appeared on the now defunct Riverside label. It was a remarkably fine disc that demonstrated Levitt's superior taste as composer and arranger and the exceptional precision of his group as performers. The same group is intact on this new Dynagroove recording, and their musicianship is on a par with their earlier triumph. In addition to five original tunes by Levitt, this new waxing includes *Oh, You Beautiful Doll, Fugue for Tinhorns, All I Do is Dream of You,* and *Don Redman's Cherry.* The sound on this platter is about the best mono jazz I've heard from Victor or anyone else. Don't miss it!

Jimmy Witherspoon: Blue Spoon
Prestige Mono 7327

This is a perfect match of vocal and instrumental forces. Prestige has supplied the ideal support for Witherspoon with a quartet made up of Gildo Mahones, piano, Kenny Burrell, guitar,

*70 Irving Place, New York, N.Y. 10003

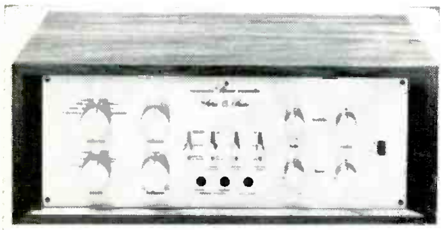
Eddie Kahn, bass, and Roy Haynes, drums. "Spoon's" melancholy is eloquently reflected in the playing of Mahones and Burrell, and the balance between voice and instruments makes this a genuine collaboration. Certainly no present-day vocalist is a more profound interpreter of the blues than Witherspoon, and this new collection gives him an opportunity to present his highly original version of *Nobody Loves You When You're Down and Out* and nine less familiar numbers: *I Wonder, It's a Low Down Dirty Shame, Back to New Orleans, It's All in the Game, Blues in the Morning, I'll Never Be Free, Once There Lived a Fool, For Old Time's Sake,* and *The Time has Come.* The recorded sound of the instrumental contribution is up to engineer Rudy Van Gelder's usual top-notch standard, but Jimmy sometimes crowds his mike too closely, creating one or two metallic sounding blasts—a minor matter when matched against the meaningful music making on this exceptional platter.

THIS MONTH'S COVER

From Herbert Cohen, of Brooklyn, N.Y., comes this attractive listening facility, which faces a fully carpeted and acoustically sound 12 × 20-ft. room. The equipment, shown below, comprises a McIntosh MR-67 tuner, (with Finco antenna and CDR rotator, and using 72-ohm coax feeder) a Thorens TD-124 turntable with an Orthophon elliptical cartridge, Crown SS-700 stereo tape recorder, Marantz model 7C stereo preamp, McIntosh MI-3 multipath indicator, two Marantz model 8 amplifiers, and two Marantz electronic crossovers, all feeding a pair of Tannoy 15-in. woofers and two JansZen electrostatic speakers, crossing over at 1000 Hz. Headphone listening is accommodated by a pair of Superex phones, and a 21-in. Zenith color TV set provides the visual entertainment. Mr. Cohen, his wife, two sons, and a daughter are all proud of the installation and derive much enjoyment from it.

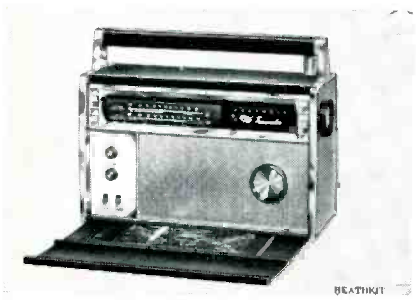
NEW PRODUCTS

• Solid-State Preamplifier. The Marantz 7T Stereo Console is the latest version of this series of preamplifiers. It is their first all-transistor unit, marking the company's entry into the solid-state product marketplace. Appearance is similar to earlier units with the addition of front-panel jacks for record and playback connection and headphones. Important specifications include: 64.5 dB of gain from phono to main output, with 42.5 dB available at the fixed-gain recording outputs. Frequency response



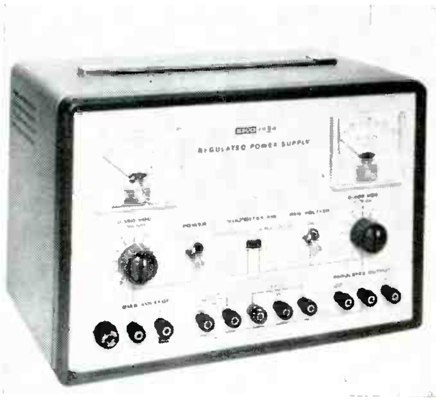
is quoted as 20-20,000 Hz.±0.1 dB. IM distortion is 0.15 per cent at 10 volts RMS output and dynamic range is stated at 100 dB above 1 μV equivalent noise input (1 μV to 100 mV at less than 0.15 per cent IM distortion). The special features of four pair of low-level inputs, dual position high-and low-cut filters, and RIAA-old 78-old Columbia LP, and tape head and microphone equalizations are retained. So are the step-switch tone controls that are out of the circuit in center position. List price of the Marantz 7T is $295.00 . Circle 205

• 10-Band All-Transistor Portable Kit. The new Heathkit GR-43 portable tunes 150-400 kHz longwave, 88-108 mHz FM, 550-1600 kHz AM, plus seven shortwave bands covering 2-22.5 mHz. Tuning a single band is made easy by a 10-position rotating dial. Assembly time is claimed as only ten hours partly because of 16 transistors, 6 diodes and 44 factory assembled and aligned radio frequency circuits. Two separate FM



and AM tuners are preassembled, ready to drop into place. The FM tuner is the same as is used in Heath deluxe component tuners. Two built-in antennas, a ferrite rod for AM and longwave, and a telescoping five-foot whip for FM and shortwave are provided. A 4 x 6-inch speaker or headphone may be used for listening. Six standard "D" cells plus one "C" cell provide power. An a.c. converter is optionally available at $6.95. Price of the kit unit is $159.95. Circle 206

• Regulated Power Supply. Experimenters in transistors are only one group that will find value in a product such as this. The EICO 1030 power supply offers two independent continuously variable sources. One provides a bias output from 0 to 150 volts at 2 mA, and the other provides 0 to 400 volts at currents up to 150 mA. Both outputs are regulated to 1/3 of 1 per cent or 0.3 volts (whichever is higher) for loads up to 100 mA. The quality of regulation is such that a change of line voltage of ±10 volts will cause less than an 0.5 volt change (maximum) in the output. In all cases ripple is claimed to be less

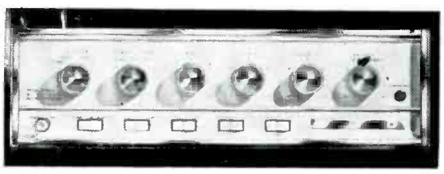


than 3 mV rms. Also provided are unregulated filament voltages of 6.3 volts with or without center tap and 12.6 volts, both at 3 amps. A front panel d.c. milliammeter and separate voltmeter can be switched for setting and monitoring either bias or regulated outputs. A convenience switch allows you to shut off power to the circuit under test without shutting off filament power to the power supply itself. This permits instant turn-on action. The supply is available as a kit or factory wired. The kit is $59.95, the wired unit is $99.95. Circle 207

• Low-Cost Amplifier Kit. Latest addition to the Lafayette kit line is this integrated amplifier. Component layout is spacious, allowing for easy construction. Three pairs of inputs are provided for magnetic phono, tuner, and auxiliary sources. The power on-off switch also incorporates a speaker phasing control; bass, treble, and volume controls are dual concentric for separate channel adjustment. Three rocker-type

panel switches select left and right channel modes and loudness compensation. There is a stereo earphone output. Important specifications include: Power output of 45 watts IHF; 3 mV phono sensitivity; 10-25,000 Hz ±0.5 dB response at 1 watt; IM distortion less than 0.3 per cent at normal listening level; hum and noise of -70 dB and -50 dB for high level and phono respectively. Output impedances for this vacuum-tube amplifier are provided for 8- and 16-ohm speakers. List price of the kit is $59.95 less cabinet. Circle 208

• Revised Transistor Amplifier. Significantly increased power output is only one of the virtues claimed for this updating of the popular Sherwood S-9000.



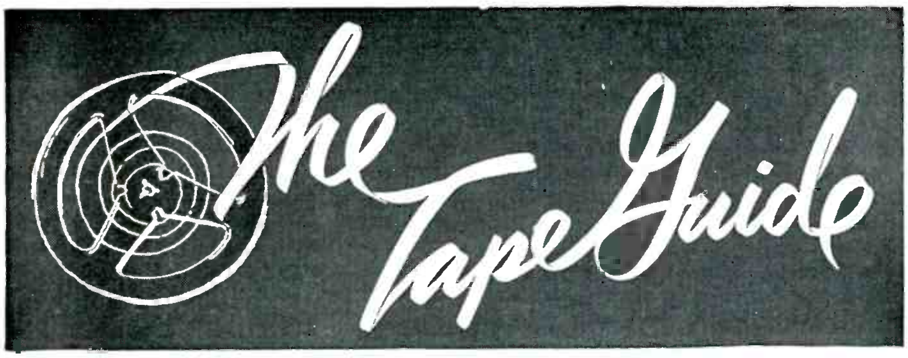
Dubbed the S-9000a, 8-ohm power output is now 80 watts IHF per channel, 60 watts continuous sine-wave with both channels operated simultaneously. Power bandwidth at 0.5 per cent THD is 12 to 25,000 Hz. Phono dynamic swing has been increased to allow for an overload point of 250 mV while sensitivity for full output is 1.8 mV. IM distortion is rated at 0.25 per cent full power and 0.1 per cent at 10 watts or less. The speaker damping factor is 40 and maximum hum and noise is quoted at -80 dB at the tuner input and -70 dB on phono. Features include front-panel rocker switches, tape monitoring, loudness compensation, and scratch and rumble filters. There is a headphone jack and a speaker muting switch. Construction uses all-silicon transistors, glass-polyester circuit boards and a regulated power supply. Size is 14" x 12½" x 4". Circle 209

• Three-Head Tape Deck. Built-in separate stereo record and play preamplifiers along with separate record, play, and erase heads allow full tape/source monitoring and true sound-on-sound recording. All solid-state circuitry is featured. This is the Sony 350 deck. Styling is in black/gold decor and the unit is set upon a walnut-grained low-profile base. Four track record and play, stereo or mono, is offered. Two VU-type meters monitor

the signals. Drive is by single motor with rim-drive rewind and take-up functions. A pause control and digital tape counter make tape indexing and editing easy. Speeds are at 7½ and 3¾ ips; frequency response is stated as 50-15,000 Hz ± 2 dB; signal-to-noise ratio is better than 50 dB; flutter and wow at the higher speed is less than 0.19 per cent and the maximum reel size is seven inches. List price is $199.50. The same unit is also available in a portable case at a list of $219.50. Circle 210

• Revised Tape Packaging. Library-type boxes for sound recording tape are now being provided by Eastman Kodak for their tape line. Designed to harmonize with room decors, the beige-colored boxes have dark brown ends and

are protected before purchase by a removable yellow sleeve. Ample space is provided on the box spine for writing in pertinent identification. The rear of the box is lined for the addition of complete notes. All Kodak 5- and 7-inch reels are now so packaged at no additional cost. Circle 211



HERMAN BURSTEIN

Send questions to:
Herman Burstein
280 Twin Lane E.
Wantagh, N. Y.
Include stamped, self-addressed envelope.

A few preliminary remarks are in order before getting to this month's questions and answers.

First, we want to call your attention to an informative and readable booklet about tape and tape recording just published by Eastman Kodak and available free of charge for single copies. It is called "Some Plain Talk from Kodak About Sound Recording Tape." Write for it to the Editorial Service Bureau, Eastman Kodak Company, 343 State Street, Rochester 4, N. Y.

Second, we would like to remind you that it is contrary to the policy of this column to make specific recommendations as to audio equipment, including tape recorders and tape, by brand and model. In other words, it is not the purpose of this column to indicate whether we think Brand A or Brand B is better, or whether Model 1 or Model 2 of Brand A is better. We can only deal with basic principles and aspects of tape recording that enable you to decide for yourself which is the best buy for your dollar and your purposes.

Third, we would like to ask you again to please enclose a stamped, self-addressed envelope with your queries. In the case of readers outside the United States, omit the stamp; the U.S. Post Office does not honor foreign stamps.

Home-Building a Tape Recorder Amplifier

Q. I am planning to construct a tape amplifier based on a magazine article. The circuit shows record equalization for 3¾ and 7½ ips, the latter being NAB equalization. However, I plan to use the amplifier with a 15-ips transport. What changes should I make for NAB record equalization at 15 ips? If you cannot give me definite values, please suggest a method of calculating these values.

A. There is no precise NAB record equalization. There is only precise playback equalization, which is the same at 15 and 7½ ips. The NAB standard provides that record equalization should be adjusted to achieve flat response when the tape is played back in accordance with NAB playback equalization. In recording at 15 ips, you will require less treble boost than at 7½ ips. The reduction will have to be experimentally determined by playing around with the values of resistance and/or capacitance in the equalization network. Thus the capacitor in the network will have to be reduced.

Adding a VU Meter

Q. I wish to incorporate a VU meter in my tape recorder. Should a cathode follower be used to isolate the meter from the record and bias oscillator circuits so as not to load them down with the low impedance of the meter?

A. It is virtually imperative to isolate the VU meter from the recording circuit, either by a transformer or by a cathode or plate follower.

Tape Ampliers

Q. I am building separate tape amplifiers for a stereo deck, each amplifier having its own oscillator. Is any form of synchronization necessary to prevent interaction (beating) between the two oscillators?

A. If you employ separate oscillators, which ordinarily is not necessary, you can synchronize them by connecting a .005-μ capacitor from the plate of one oscillator (either plate in the case of a push-pull oscillator) to the corresponding plate of the other oscillator.

Series-Parallel Heaters

Q. I notice that in a number of audio components where d.c. is applied to the heaters, all the heaters are in series. Could not the individual heaters be arranged in series-parallel?

A. It is possible to employ a series-parallel arrangement in order to facilitate use of an available d.c. supply. However, there are two reasons for putting *all* the tubes in series: (1) If the heater of one of the tubes becomes shorted, the resulting voltage overload on the other tubes is reduced to the extent of the number of tubes in series. For example, if four tubes share 24 volts, a short in one tube will result in 8 volts across each of the remaining three. But if two tubes share 12 volts, a short in one will result in 12 volts across the other. (2) The higher the voltage of the supply, the easier it is to filter it. Thus much larger values of filter capacitance are required for a 12-volt supply than for a 24-volt one.

Quality Degradation

Q. When my tape recorder was new at 7.5 ips I could not tell the original and playback signals apart, and at 3.75 ips the difference was barely noticeable. After a year I notice a decrease in quality. Reproduction at 7.5 ips is barely tolerable by hi-fi standards; it is quite poor at 3.75 ips. When this trouble started, I checked head alignment, pressure pads, bias, and so on. I noticed that the bias voltage varied from night to night, partly causing the trouble. But even when the bias is optimized the machine does not record as well as when new. I do not think the playback head is at fault, because old tapes recorded when the machine was new still sound all right. What, besides variation in my house current, can cause the bias to vary?

A. If you haven't already done so, among the first measures you should take to assure extended treble response are head demagnetization and head cleaning. The bias variation may be due in some part to the oscillator tube. Have you tried replacing it? It may be due to the capacitors employed in the oscillator circuit. They should be mica (preferably silver mica) rather than ceramic or paper. Possibly the electromagnetic characteristics of the record head have changed, perhaps because of a shorted turn, physical shock, or the like. Therefore replacement of the record head, although inconvenient and not inexpensive, is an indicated measure. I take it that your machine employs separate record and playback heads; have you checked the azimuth alignment of the record head relative to the playback head?

Alignment Tapes

Q. In one of your writings you mention the use of the RCA track alignment tape No. 12-5-64T, for quarter-track heads. I have been searching unsuccessfully for this tape. Where is this tape available?

A. I suggest that you write to RCA Victor Record Division, 6800 E. 30th St., Indianapolis, Indiana. At least that is the information given me by RCA in New York City.

Mono Usage of Four Track Tape

Q. I have been warned not to tape monophonically on all four tracks when using a quarter-track machine because of inevitable cross-talk. Is this warning justified?

A. My experience with top quality tape recorders is that one can record successfully on a quarter-track mono basis, with crosstalk being virtually non-existent and therefore not a real problem. With machines of lesser quality, crosstalk does sometimes become great enough to be objectionable. One of the qualities of a high-grade stereo head that distinguishes it from a run-of-the-mill stereo head is the minimization of crosstalk within the head. The same is true for high-grade tape amplifiers *vs.* ordinary ones. Thus you can see where some of the money goes in the better tape recorders.

Bias Traps

Q. I have a professional stereo tape recorder that records and plays simultaneously, and in checking the harmonic distortion of each channel at 400 Hz I obtained readings of about 26 per cent, which, of course, is much too high. I believe this is due to the fact that I was measuring the bias frequency as well as harmonics of the audio-signal. Some time ago you gave a formula for determining the capacitance that is resonant with a given inductance at a given frequency, with inductance stated in millihenries and frequency in KHz: $C = 25{,}000{,}000/f^2L$*. Inasmuch as the bias frequency of my tape recorder is specified as 68KHz, and using a 2.5 mH inductance, I calculate that the required capacitance for a bias trap I wish to construct is 2162 pf. Is this correct?*

A. This is correct. But the easiest way to tune the trap exactly to the oscillator frequency is to use a variable inductance, such as a TV width coil variable between approximately 1 and 10 mH, along with a fixed capacitance of about 2000 pf. I take it that your bias trap will consist of the capacitance and inductance in series connected between the audio output and ground.

Bias Voltage Change

*Q. I have a **** stereo tape recorder which includes a VU meter that can be switched to read bias. The bias stays constant in the right channel but drops about ½ dB in the left channel during the first 30 minutes. I have changed the bias oscillator tube, but with no improvement. Does the decrease in bias degrade the recording?*

A. A change of about ½ dB in bias current will have a slight to moderate effect, depending in part on your starting point; that is, depending on whether the bias is initially at the optimum value or slightly greater than optimum. Treble response will be somewhat elevated, and distortion somewhat increased. The effect also depends in part upon the particular tape that you use. Some tapes are more tolerant—have a sensitivity curve with a broader peak—than others with respect to a variation in bias. Finally, the effect will depend in part upon tape speed; the higher the speed, probably the less noticeable will be the effect.

Try changing the variable capacitor that adjusts the amount of bias current to the left channel. Æ

LIGHT LISTENING

(*from page 8*)

divert the ear from the echo chamber and forced treble so common in today's popular releases. This, of course, is not the sort of sound found in all Columbia releases. A classical Columbia tape issued at the same time as this Faith reel (Vivaldi's Four Seasons with Bernstein and the New York Philharmonic, MQ 736) also features strings in no uncertain fashion. Unlike Faith's recording, it delivers the natural string sound that one can take for granted these days. The Faith reel begins to sound like the Bernstein only when treble is steeply rolled off at about 10 kHz. Æ

AUDIO ETC.

(*from page 12*)

performance. I might note, just for info, that the pickup also comes in standard-adapter form, for any old arm, including changers, offering the same two basic cartridges (bi-radial and plain conical diamond) and that the small lightweight power supply now necessary for their use is correctly plugged and valued, with HI output direct to an AUX input, by-passing preamps, and for convenience a LO output with "re-equalization" to overcome the standard magnetic-input equalization—this last not really recommended with much conviction by the company.

In an oddly undefinable way, this unit does in fact suggest a ceramic in actual operation—a superceramic—and of course offers such happy ceramic qualities as total lack of hum, over-all hi-level signal and the ultimately desirable absence of preamplifier. I do not know how thoroughly the Euphonics people have got their system protected—but it does seem to me that here, within the ceramic family of pickups, is a true cartridge-for-the-future which, eventually, might replace most of the magnetics and ceramics of present designs—and in the process eliminate the pesky preamp, which could easily stand elimination if its services were no longer needed.

By the usual technological process of competitive practical improvement, the magnetic is decidedly king for today and maybe tomorrow too. It's that good. The new Euphonics type isn't for the future because it sounds "better" by any huge degree—though its small increments of improvement are definitely important—but simply on the basis of the *system itself*—the solid-state micro-valve that frees the stylus of its generator burden. That is something we can't put aside.

MICROCIRCUITS—THE HI-FI KNOB

On my desk, each month, lie copies of my two favorite science magazines—AUDIO and *Scientific American*. In one of them I write about whatever I may know. In the other I read and gasp.

I suppose I might gasp once in awhile at the marvels disclosed month by month in our own magazine. But mostly, I've already heard about them, and so there would be no point in gasping. But in the other magazine, which covers the whole large area of science, I'm a babe in the woods, innocent and unknowing—and I'll bet a hat you'd be the same, for the most part. The coverage there ranges over areas as far apart as archaeology and genetics, with vast quantities of chemical, biological, and electronic data in between. Electronics, of course, turns up in almost every article in some fashion. But once in awhile there is something directly concerned with electronics itself—or themselves. In that case, the spotlight hits close to home—*our* home. AUDIO.

Take the November, 1965, issue of that magazine. (It was just in, when I had to meet this particular deadline.) The cover itself is one of those things to make an audio buff gasp. *Microelectronics* is the title, and the handsome art work, in turquoise blue with maroon line drawing, depicts a circuit. What a circuit! It contains no less than 22 transistors and 46 resistors and it is an actual, working circuit, not a diagram. On the cover it is a big square, seven inches on a side, full of complex lines and channels and divisions. And yet, do you know how big this actual circuit-module is in the flesh? *Less than a sixteenth of an inch square*, on a silicon wafer. They can turn out hundreds of thousands of them and others of the sort, to the most complex circuit specifications, these days.

Sure, no surprise for a lot of you who are up to date on this sort of thing. But you must pardon me for getting het up. For this really hit me—the long, detailed article inside the magazine, which goes into the whole field of microelectronic integrated circuits as of the present. I'm reeling, trying to bring it all into focus with the electronic aspect of our own business, as of the years to come.

Not that I haven't heard of microelectronics of this sort. I'm not that dumb. We all know about the marvelous new etched and deposited miracles, at least insofar as transistors themselves are concerned, the individual units. And also in connection with the many micro-scale circuits, of the printed sort and related types, which we already use in and out of audio. But here is an advanced story, written up complete in black and white (and turquoise and maroon), to smack the eye, some ten pages of text, photos, graphs, and diagrams that make the whole process of turning out complete integrated microcircuits ultraclear for the goggle-eyed initiate, including designs, manufacturing techniques, the different types of approach, and the new applications for micromodule electronics in a vast range of practice. Enough to make anybody gasp.

10,000,000 components

There's an amusing graph, in this article, entitled "Growth in Electronic Complexity," on which the vertical variable is the Component Count per System (ugh—let's just say the number of transistors, diodes, resistors, etc etc.) and the horizontal is a time-scale, running in five-year units from 1950 to 1970. The "graph line" is merely words, spaced out on the graph space.

At the left bottom corner is "Hearing Aid." It had a very few components, as of around 1950. At the upper right is "ESS No. 1," an electronic switching system upcoming from Bell, as of after 1965. In between, on the scale, are Portable Radio, Television Set, 1960 Computer, Telstar, 7090 Computer—and then that ESS No. 1 thing. What intrigued me was the component-per-system vertical scale itself, which is measured in log numbers, 10, 10^2, 10^3 and on up to 10^7. That's 10,000,000 if I log rightly, i.e., 10,000,000 electronic components for a single "system"! They come that way, or will soon.

Of course the average home entertainment circuit contains a goodly number of components, by our own standards, and we are all aware that the home TV set in this respect takes a very small cake of sorts. It *looks* pretty complicated inside, as any serviceman will say. Indeed, on this graph, we note that the portable radio lies just under the 10^2 line—with nearly 100 separate electronic components —and TV is more than halfway up to 10^3, as of around 1955 (when the present sets began to crystallize out, so to speak). So home TV already involves us, per unit, in something along towards 1000 components, counting only the electronic elements.

It is obvious that even though tiny size is helpful when microcircuit compound component units are used in such a modest numerical situation, even more important is, first, low-cost mass production of the miniature circuit elements and, second. Their new *reliability*. For the higher up we go in the scale of componentry numbers, the more astonomical are the chances for chaos through failure. Not to mention the little problem of locating the failure.

The 6600 computer made by the Control Data Corp., for instance, sits 'way up near the top on the *Scientific American* graph just below the 10^6 line, as of 1965. The text elucidates: this machine contains approximately 100,000 diodes, 350,000 transistors and 500,000 resistors! And (it says) they are all individually assembled. On the same basis, Telstar isn't far behind—and you can count in all the other space machines and telemetering services and whatnot now flying, or sitting on the ground, monitoring. Fantastic.

Hence in these 'way out fields there has been a fantastic pressure to develop circuit elements both more compact and more reliable than any *individual* component can ever hope to be, in wired connection with a couple of thousand or a million others. The integrated, one-piece microcomponent circuit is the result. And the field is booming at an absolutely thunderous rate. As *Sci. Am.* succinctly puts it, in the account of the cover picture, "Increasingly . . . these tiny objects, known as integrated circuits, are being adapted for nonmilitary computers *and other electronic systems* (italics mine) because of their low cost, high performance and *extraordinary reliability* (italics mine)."

Graduated Increments

Well, there you have it. The question is, how will this astonishing development affect us, eventually, in our own little microcorner of the big world of electronics? How can we best put these minor miracles to use in the relatively prosaic business of sound-making? (Purely relative; we're plenty glamorous, in our own special way!)

Well, you may be sure that plenty of audio men are thinking about this, and with a more practical approach than can ever be mine. Practicality isn't my biz. I'm all for wild speculation about how things just *might* turn out in the future. (If I turn out to be right, then I'm practical after all.) But, speculation or no, I can say one thing quick. No business like ours moves towards successful change except in graduated increments. It just ain't possible, nor wise. Home-type gadgetry must be proved out in the home itself, after sale, and no amount of pre-sale "laboratory testing" can jump us forward, say, ten years at a jump, to bring out 1976 hi-fi next fall. So let's not get too quickly excited by the possibilities for complete amplifiers that are a sixteenth of an inch square. Though it could be.

I always remember, in this connection, the example of the Chrysler Airflow cars back in the middle thirties. That was a deliberate attempt, at least in styling, to jump far forward. The cars did exert an immense influence on later design and still do. They were sensational in that day—do I remember old near-sighted Alexander Woolcott peering short-sightedly out of that unbelievably rakish slanted rear side window! But, even so, the actual cars themselves, as we now can see with hindsight, were remarkably clumsy and gawky affairs, half-baked mix-ups, combining brilliant innovation with ugly bad-guessing in a design that doesn't stand, so to speak, on its own four wheels. You just can't jump ahead that fast—*wholly*, uniformly, ahead. It never works. You end up with a monster.

And so we won't have grain-of-salt amplifiers quite yet.

And yet—it makes one think. Can't help it. All you have to do is to read that *Scientific American* article on the details, the *practical* details, of the new microcircuit techniques, to get all excited. For here you have the whole thing already a reality, with large clear pictures of manufacturing techniques, block diagrams explaining the production stages, the silicon crystals, the growing-on of N-type layers oxidizing, masking and etching, oxidizing again, adding new layers, and so on, until in one tiny block the whole latticework of components is built into a solid, single unit. Just extraordinary, and no two ways about it.

I am happy, therefore, to plug my other favorite magazine for all I'm worth. If you want to read all about it, go right out and find the November, 1965, *Scientific American* (with the microcircuit cover). Or send for an offprint, at 20¢, to W. H. Freeman and Co., 660 Market St., San Francisco, Calif. 91404. Might as well order 5 and enclose a dollar.

Hands and Fingers

It's obvious that we are never going to have to cope with electronic components in the hundreds of thousands per hi-fi "component." (We're going to have to watch that word.) Nor do we have the enormous space-technology problem of maximum component content per unit of weight and size. We in hi-fi do *not* need to cram 100,000 components into a couple of square inches and a few ounces of total weight. We never will.

We are limited in our moves towards compactness by such elemental things as hands and fingers, which don't get any smaller as time rolls on. Who wants an amplifier so small that you can hardly get hold of it? And what do you do about the control knobs?? And the connecting wires???

Then there's the loudspeaker, that irreducibly large(?) transducer. The input end of sound reproduction is still, shall we say, somewhat fluid, in flux, as to the size and weight of its major components and assemblies, with new developments still coming along—though not without problems. But at the speaker end, we have come down flat to at least two seeming irreducibles. On the one hand we have the cone and diaphragm "bookshelf" speaker and on the other the electrostatic. Both can be made in many sizes and shapes, with correspondingly variable range of frequency coverage. But the *ratio* of size to bandwidth remains roughly constant. We have bogged down in sidewise motions, recently, not really moving forward in respect to more bass for a given space or, alternatively, less space for a given bass.

I'm talking in extremes—in terms that might match up to the microcircuit development in electronics. We are miles away from a microspeaker (except headphones) to match the size-potential of the electronic elements now appearing. For that, we'd want a full-range unit maybe an inch across and a half-inch deep! Fat chance. The only place where we have really begun to operate in this area is in the littlest transistor radios—but at a slight (!) sacrifice in tonal range, alas.

Puff, puff, puff,

Still, and again . . . (Here I go.) You *could* (I'm thinking out loud . . .) put all your hi-fi components, your electronic elements, into a thumb-sized box and hook them up to a standard speaker at one end and a record player at the other. Or mount the tiny box inconspicuously on either unit. It could nicely contain an amplifier, stereo, plus a complete stereo FM tuner and a preamp, if desired. But those nasty big wires? Maybe you could add, also inside the same little box, a high quality UHF (for a short antenna) broadcast wireless unit, and put a corresponding receiver (40 transistors, 100 resistors and capacitors?) in another little inch-square remote container. Look, Ma, no wires. Just a pair of sugar-lump squares.

You'd have to attach the loose amplifier-receiver square *onto* something (I'm thinking), or it might get lost, or sucked into the vacuum cleaner. How about building it into a table lamp? Or a book end? Or an ash tray? Or encapsulate the whole unit in one of those handsome clear-plastic cubes? (But look—that's making it *bigger* again. Ugh.)

Nope, this won't do. I forgot all about the control knobs.

OK, then, let's build a remote control—into the same little box. Lots of room left! You just blow on the thing. One puff and the volume goes up. Two puffs and it goes down. Three puffs and the highs are boosted 6 dB at 10^4 Hz. No knobs. But who has so much breath?

(*Dear ETC: Let us have a puff on that before you throw it away.* ED.)

Forget it! The greatest use we'll be making of these remarkable tiny one-piece circuits in the future will be to take advantage not so much of their size as their low cost and their extreme reliability. That's common sense. With them, eventually, we'll have more versatile, more reliable hi-fi for the money. An RMS guarantee for 20 years, unconditional—could be.

We have actually gone a long way in this direction with present solid state.

We see it already today, the more-fi-in-less-space, the bigger-sound-for-less-money, in many a 1966 model. Including, for a quick example, E-V's 1178 stereo amplifier-receiver on our November back cover, which is "no taller than a coffee cup" and "no bigger than an open book." It's much, *much* bigger than its theoretical microcircuit electronic equivalent, which would probably take up a square inch of space. But the E-V, nevertheless, is a lot smaller than, say, the nearest 1955 equivalent, mono only. Really big difference. Increments, again. Good ones.

And remember, the E-V 1178 has knobs on it, for fingers. They really are awfully handy, when you come down to it.

*****

I've got it! The Ultimate Answer. Mount all your hi-fi electronic microcircuits *into the knobs themselves.* Then just fasten the knobs onto something . . . er, well, say a nice hunk of solid oak. Better still, screw them down onto the arm of your favorite easy chair. Or make 'em removable, and "plug them in" any old place where you need them, all over the house. What an idea! The Hi-Fi Knob. Seventeen models to choose from, seventeen complete Systems . . . Æ

JUNK BOX METERS

(*from page 26*)

dicators. The full-wave a.c. voltmeter circuit of *Fig.* 2-5 may be used as a volume-unit (VU) indicator for this purpose.

An actual VU meter is a well-defined, standardized instrument.[3] It uses a copper-oxide rectifier, has a specified internal impedance (3900 ohms), a certain response time (0.3 second for 99 per cent deflection), and is calibrated for use only on a 600-ohm line (indicates 0 VU for 2.5 milliwatt of power at 1000 Hz).[4] The average home experimenter probably does not need all that precision, but merely an indication of relative signal amplitudes. In that case, one of the circuits shown in *Fig.* 2-8 should suffice.

Circuit (A) is recommended by Hogan[5] for addition to a vacuum-tube tape recorder having approximately one volt rms audio signal level available. Circuit (B) was used by Abajian[6] in a tape recorder having less than 100 mv of signal. Both circuits were designed for a standard VU meter, but should work satisfactorily with the full-wave a.c. voltmeter circuit of *Fig.* 2-3. For a close approximation, use a meter movement of 200 μA full scale and 3900 ohms resistance. Otherwise use any meter from 50 μA to 1 mA and add series resistance if necessary for a total of approximately 4000 ohms.

Mark the meter scale for zero VU or "normal" at 0.71 of full scale (71 per cent deflection). Calibrate the circuit as follows:

1. Apply a 1000-Hz tone to the input of the recorder and increase the amplitude in steps until distortion becomes noticeable on playback.
2. Reduce the input signal by 12 db, *i.e.*, to one fourth of its original value.
3. Adjust the VU meter circuit to indicate 0 VU at this level of the input signal.

This adjustment depends a lot on the ballistics of the particular meter used, and some experimentation may be necessary to find the proper setting. The basic idea is to set the meter to indicate low to compensate for the occasional audio peaks which could drive the recorder into distortion before the meter has a chance to respond.

Summary

The requirement for a meter to be installed in a piece of home-built equipment is often unsatisfied for lack of funds to purchase a special instrument. This article has shown how to adapt a junkbox meter to measure direct and alternating currents and to perform several special functions such as measuring resistance or audio signal levels. Now drag out those dusty meters and complete your long-neglected projects!

—Æ

[3] "*Radiotron Designer's Handbook*," edited by F. Langford-Smith, Harrison, N.J.; Radio Corporation of America, 1953, page 823.

[4] 0 VU on the scale of a standard VU meter, with a 3600-ohm resistor in series with the meter, and across a terminated 600-ohm line is actually +4 VU, or 2.5 mw. The *reference* level is 1 mw, but the meter is not sensitive enough to indicate zero at that level. ED.

[5] J. W. Hogan, Adding VU meter to tape recorder, *Electronics World*, October, 1964.

[6] Dr. J. Abajian and D. V. Jones, All-silicon tape-playback preamps and recording circuit, AUDIO, October, 1964.

EQUIPMENT PROFILE

(from page 50)

HARMAN-KARDON SC-440 COMPACT STEREO SYSTEM

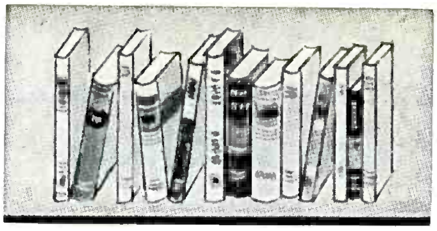
While the average audio buff may not consider that compact music systems are the equal of his $2000 component system, we cannot overlook them. They *are* popular; they are not as expensive as our regular systems; they fill a need for homes where there is not enough room for two large speaker systems which came in boxes which could serve as Volkswagen garages. Besides that, some of them offer exceptionally good listening. In addition, they often provide convenience features not otherwise obtainable.

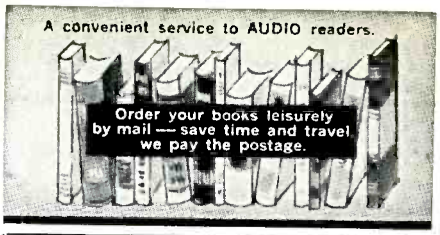
One such system is the Harman-Kardon SC-440 which was introduced only last fall, and since has chalked up an impressive reception in the marketplaces. It consists of three units—the receiver/record-player unit, and two loudspeaker systems. The first part consists of 5-in. high cabinet with its control panel at an angle of about 30 deg. from the horizontal, and with the Garrard AT-60 record changer flush-mounted on the top with its usual spring suspension. An optional plastic dust cover is available, increasing the over-all height to 9 in. This cabinet arrangement makes it possible to put the unit on even a low table and yet have the controls and the tuner dial readily visible. The loudspeaker systems are rectangular "bookshelf" units, though of sufficiently generous size to provide adequate bass response. They actually measure 22⅜ in. high, 13⅜ in. wide, and 10½ in. deep. All of the wood surfaces in the entire system except the speaker cabinet backs are finished in oiled walnut of a particularly pleasing, fairly light shade. The speakers consist of a 10-in. high-compliance, long-excursion cone woofer and a 3½-in. cone tweeter in its own acoustic housing. A carefully designed crossover network provides a response which is smooth throughout the crossover region, and the separate tweeter housing prevents back pressure from the woofer breaking up the tweeter—a problem in some systems which do not have this feature. The crossover is in the vicinity of 1250 Hz; response is definitely measurable to over 18,000 Hz, and fairly uniform down to 50 Hz and falling off somewhat below that, although still audible at 30. At 10,000 Hz, response 45 deg. off axis is down only 5 dB from on-axis level.

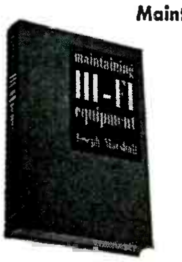
One intriguing feature we have not encountered before is the removable grille with the simplest of fastening devices—six little pads of nylon "hook" fabric are mounted on the baffle, and six more on the grille panel. The latter is held on firmly, and with no metal-to-metal contact there is no chance of rattle developing.

The Garrard AT-60 is so well known, of course, that little has to be said about it. A groove routed on the wood cabinet behind the turntable serves to hold the unused spindle.

The control panel mounts a selector switch, a tuning knob, power switch/volume control, balance control, bass and treble controls, and two slide switches—one to add contour, and the other to cut off the speakers. There is, in addition, a headphone jack, tuning meter, and stereo indicator light, and, of course, the AM/FM tuning dial, softly illuminated in green. On the rear of the cabinet are the speaker pin-jacks, aux input jacks, and recorder-feed jacks. A 4-ft. hank of wire is attached to one terminal of the FM antenna strip, which will accommodate a 300-ohm dipole. Until we started checking performance, we did not even unfold the hank of wire, so the unit was performing on an effective 4-in. length of wire, and still limiting on all the N.Y. City stations. The AM antenna is a loopstick built inside the case.







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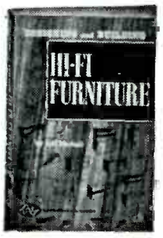

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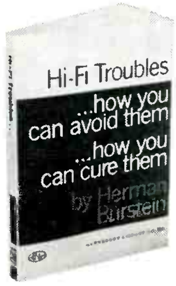

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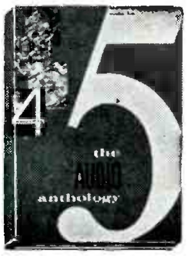

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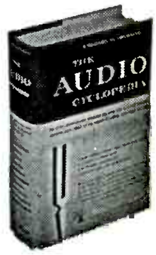

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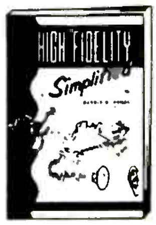

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Specifications

While technical specifications are rarely stressed in either literature or advertising for the so-called compact systems, the true audio buff still remains interested in them. The SC-440 is a "system" version of the SR-400 model in the Stratasonic series. It has an IHF output of 36 watts (both channels) at a distortion level of 1 per cent. Frequency response is ±1dB from 15 to 25,000 Hz at 1 watt, or 20 to 23,000 Hz at full power. S/N is 90 dB, and the damping factor is 25. The FM tuner has a usable sensitivity of 2.9μv, and on AM the sensitivity is 50μv/meter. In all these departments the specifications are the same as for the SR-400. But what importance the specifications are with respect to a complete system remains to be determined. What *is* important is the over-all sound quality, the convenience of handling, and the quality of the loudspeaker systems. Our first experience with this model was at the Music Show last summer, when we were standing near one speaker when it gave out with a particularly impressive drum passage which came as a complete surprise—it was that good. Both in operating convenience and physical attractiveness, the SC-440 stands high in our estimation, it is attractive, and it is a good performer.

Circle 204

RECORD REVUE

(from page 47)

voices too, but what is most evident is the interpretation—the "Pro Musica" style, now so familiar in dozens of later recordings and thousands of concerts. (These are early recordings, I think.) The Pro Musica singing here is typically brilliant, hard as nails, showy, fast and rigidly metrical—it bangs along in strict time with never a trace of word-shaping or plasticity, except as one or another singer can't help it, now and then, on his own,

Admittedly, this style of performance has been hugely successful—and it is both brilliant and persuasive, But for my slightly pained ear it just isn't right, nor musical. Everything sounds alike; sensitivity, musical "poetry," is replaced by vocal and rhythmic efficiency, loud, brassy, hardboiled. Try for yourself!

Telemann: Four Sonatas for Flute and Continuo ("Methodical" Sonatas). Samuel Baron, flute; A. Kouguell, cello, R. Conant, hps.
Dover HCR-ST-7004 stereo

Not all the Dover low-price line is mono and/or reissue; this is new compatible stereo, adding a domestic performance of Telemann to the numerous European offerings now available

It's good, too, though for some reason Samuel Baron's flute has a dullish sound in the recording. There is a minimum of "New York" hardness here, and plenty of good Baroque communication to match anything from the continent.

The Sonatas are "methodical" in the sense of a "method"—they were pedagogical examples of good composition and, in particular, were published with the slow-movement ornamentation completely written out, whereas ordinarily it was left to the performer to improvise. Thus they are invaluable guides to performance practice. Yet like many "teaching" works by big composers, these make first-rate musical listening as well, graceful music in the characteristic Telemann vein—a sweeter, more fluent Bach, less profound but easier, more pliant.

Æ

SPLIT-LOAD STAGE

(from page 21)

Out of curiosity I have done the simple sum of examining the design of this transformer. The inductance must be worked out for the drive conditions, not the low-level response conditions, and if we allow ourselves to lose 3 db of drive at 30 Hz, we must have $200 L = 10$ ohms, where L is the half-secondary inductance and 10 ohms is the power transistor input resistance. On a small transformer we might expect to get 1H/1000T, as a round number, so that 1/20 H will need about 225 turns. If we calculate for response alone we find ourselves using 5 mH instead of 50 mH but the low-impedance generator must then be prepared to supply the very high magnetizing current at 30 Hz and the core must be prepared to stand the very high flux density. This is a well-known trap in the design of audio power transformer, a trap which can catch the designer or the customer who relies on a frequency response measured at low levels.

The result obtained in the discussion of *Fig.* 6 is sufficiently surprising to merit a closer examination, for I must confess that I find it rather hard to believe. Surely the split-load circuit cannot give us a lower impedance than the emitter follower. In order to clarify this point we must consider the essential circuit shown in *Fig.* 7. When we look back at the primary of the transformer, which is what we are doing in *Figs.* 4 and 5, we see the impedance $(1 + n)/g_m$ and this, of course, gets larger and larger as we increase n from its emitter-follower value of zero. If we retain the secondary at unity instead of allowing it to follow the $(1 + n)$ movement, we get for the impedance which loads the transistor the value $(1 + n)^2 R_l$ and for the source impedance we get $(1 + n/g_m)/(1 + n)^2$. In fact the source impedance is $1/g_m\ (1 + n)$ and thus if $n = 1$ and $g_m = 1A/V$ we get the value already quoted, ½ ohm.

Improvement in Working Conditions

The price we have paid is quite simple: we have increased the voltage swing and reduced the current swing. In a driver situation this is all to the good, because we will almost always have plenty of voltage and a serious current limitation. The power transistor may need 50 mA, 100 mA, 200 mA or more in the way of peak base-current drive, leaving out the transient conditions, but it will rarely require a volt at the base to produce this current. The drive transistor will be fed from the common supply line at 12-14 volts, or 24-28 volts and if we use an emitter follower we must get rid of most of these volts in order to reduce dissipation.

Just to do some arithmetic let us assume that we set our driver to operate at 40 mA, 10 volts, which gives us 400 mW dissipation. By making $n = 9$ we can provide a 1-volt, 400-mA drive to each power transistor from a source impedance of 0.1 ohm and we can keep the common-base drive needed for the transient up to 400 mA instead of the emitter follower 40 mA.

You notice that what we are doing is what we might have done when we were very young. We look for the optimum load impedance for the driver transistor, we design the circuit to fit this to the load, we use negative feedback to bring down the source impedance and to increase the input impedance and we reach our solution. By using this logical approach we have created one new problem. The driver load is 250 ohms and so we must make our primary inductance something of the order of 1 H or more for our 30-Hz response. We shall find ourselves back in the transformer design business. I do not feel too worried about this. In the first place, if we are going to use a transformer we should be prepared to sweat out a design: in the second place I do not think the design problems are necessarily too difficult. We shall have a fair number of turns on the collector winding and we shall need a good air gap to deal with the polarizing current. The transformer will not be a very small one. We have, however, something which we can call a feedback winding, the emitter winding, which has the same number of turns as each half-secondary. These three windings can therefore be a trifilar system and we shall have very tight coupling between the secondary and the feedback winding. Problems of leakage inductance will be greatly reduced and I feel that it is only the sheer physical size of the transformer which can be counted against it, even though it will still be small compared with the transistor heat sinks.

It remains for the reader to consider whether he would prefer to get his low impedance by feeding back around two stages, by using positive feedback, or however. Here we have simply arrived, I think, at the conclusion that if you use a transformer at all, you will do well to let it transform and not simply use it as a phase splitter. This is to choose the worst of two worlds. Æ

AUDIO MEASUREMENTS

(from page 37)

related to another characteristic of the program material and called by the related name:

Transient Response. All the tests mentioned so far are performed with steady tones: the response of the amplifier or system to steady tones of various frequency, for frequency and power response; the generation of harmonics by a single steady tone, or of intermodulation products by more than one steady tone; and noise in the steady absence of "signal." But program material contains frequencies that come and go. When they come, or start, relatively suddenly, some observers noted distortions that weren't noticeable with steady tones. So the realization of transient distortion was born.

Many tests have been proposed and have gained some acceptance, for detecting different forms of transient distortion, most noteworthy of which are square waves and tone-burst testing. The kind of test waveform for these are shown in *Fig.* 1-9. We'll describe them, as well as discussing their relative merit, when we get to this part of the subject area.

That's run the gamut of basic measurements encountered in the simplest of amplifiers or systems. As we proceed to other items in a system, we shall encounter other measurements that specifically apply to those items, such as equalizers and preamplifiers or control centers, phonograph pickups and turntables, tape recorders and tuners, microphones and loudspeakers.

In our next installment, we shall start going into detail about these basic amplifier measurements, beginning with frequency response. To get you thinking about this, look at *Fig.* 1-10. Is this all that is needed to measure frequency response? Compare your conclusions with our own, given in the panel.

OUR Conclusions

First error in the set-up of Fig. 1-10 is the implicit assumption that the audio oscillator gives the same output at all frequencies—no provision is made to measure it. So a meter should be included to measure input as well as output. Next, it is highly probable, specially with high-gain amplifiers, that the oscillator will give far too much output, overloading the amplifier before you even start to turn up the oscillator output control, possibly on hum from the oscillator. So you need some form of attenuator to cut down the oscillator to a level usable for the amplifier input.

Further omissions are consideration as to whether the input and output of the amplifier have the correct terminating impedances—input source and output load. We haven't mentioned them yet, have we? Based on these answers, suppose we revise our notion of what is required to the set-up shown at *Fig.* 1-11: is this adequate? Seems as if it should be, doesn't it? But how do you know that the meters both read voltage correctly at all frequencies? Suppose they don't, is there any way of making sure the measured response is still correct? And have we shown the output meter connected on the correct side of the input source resistor?

We'll answer these questions in our next installment as well as showing how frequency response can be automated, or improvised with limited equipment. In later installments we'll proceed to other amplifier measurements, as space and time allow.

ABOUT MUSIC

(from page 14)

session, during which Miss Laretei obligingly repeated the same movements many times. Complimented on her patience, she said, "Oh, I'm used to this. Rehearsals on television are much more tiring. Here it's the sound you're after, on television the lighting takes so much more time."

Commenting on the first playback, Miss Laretei said: "It strikes just the right balance between intimacy and brilliance. But what especially pleases me is that I can hear my performance out there. Engineers always tell me: 'Don't worry about how it sounds to you. What counts is how it's being recorded.' But I cannot work at my best that way. I *must* hear myself; otherwise I feel I am playing into a vacuum. In this studio, it's lovely." Looking at the crimson rugs, she added, "And such elegant acoustical damping!"

In the concert hall, before the television cameras, or in the recording studio, Käbi Laretei is a musical missionary of rare charm and persuasiveness. She could be one answer to what one critic recently called "the dying recital." Æ

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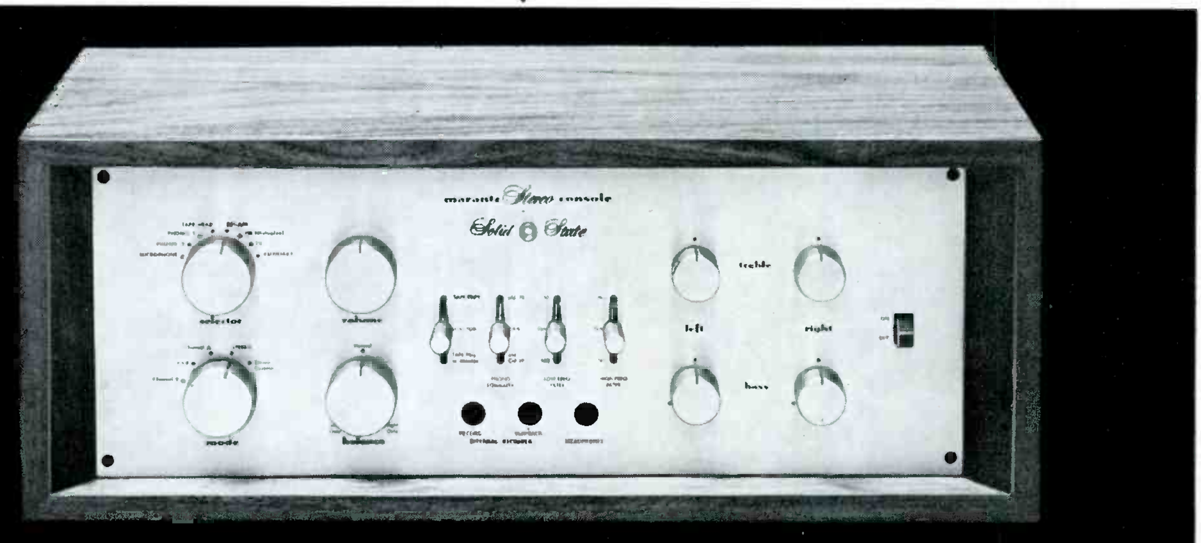

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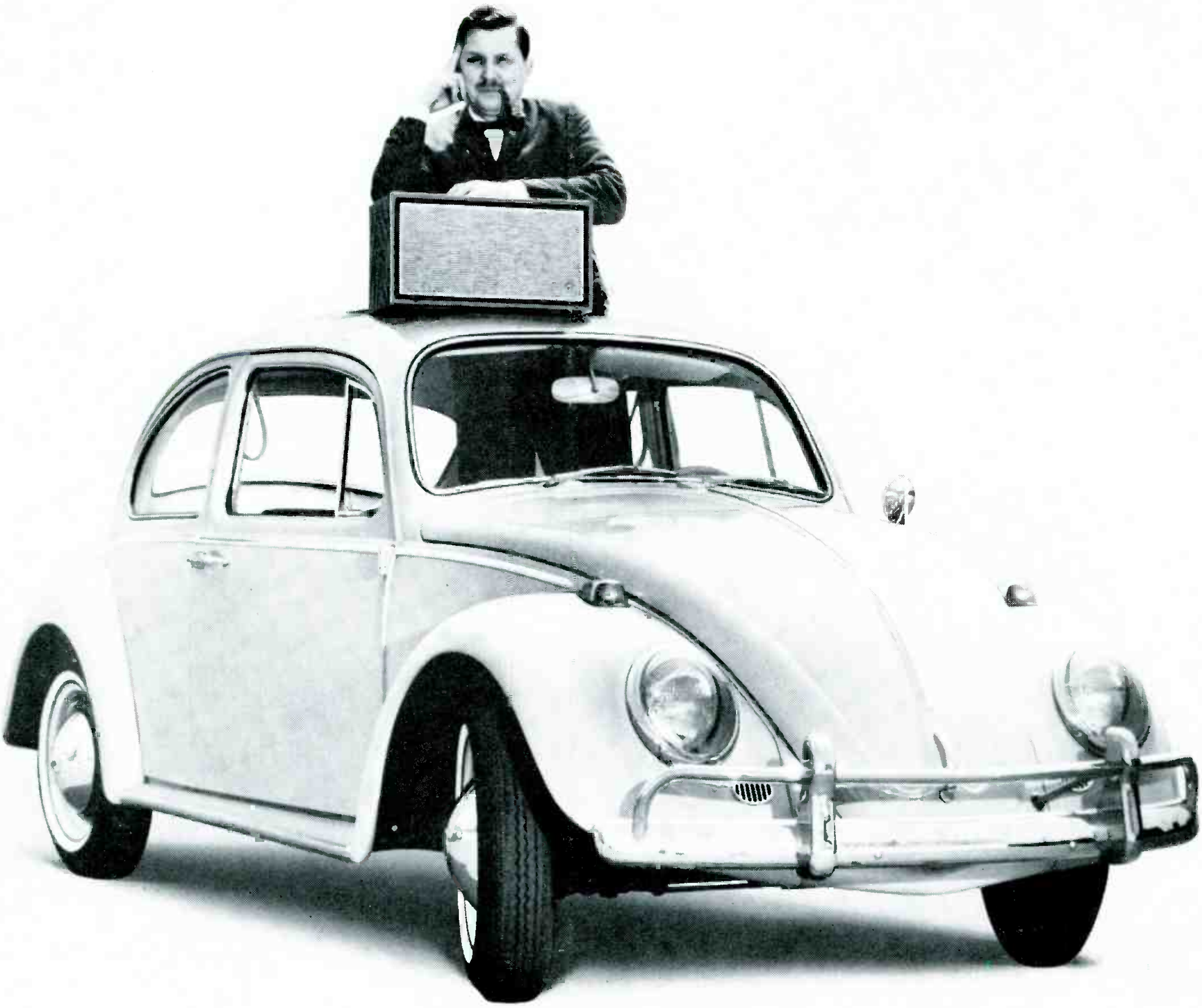

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