the authoritative magazine about high fidelity

AUDIO

DECEMBER 1968

60¢

THE 20 MOST-OFTEN-ASKED QUESTIONS ABOUT "HI-FI" (WITH ANSWERS)

LOUDSPEAKERS vs. MUSICAL INSTRUMENTS

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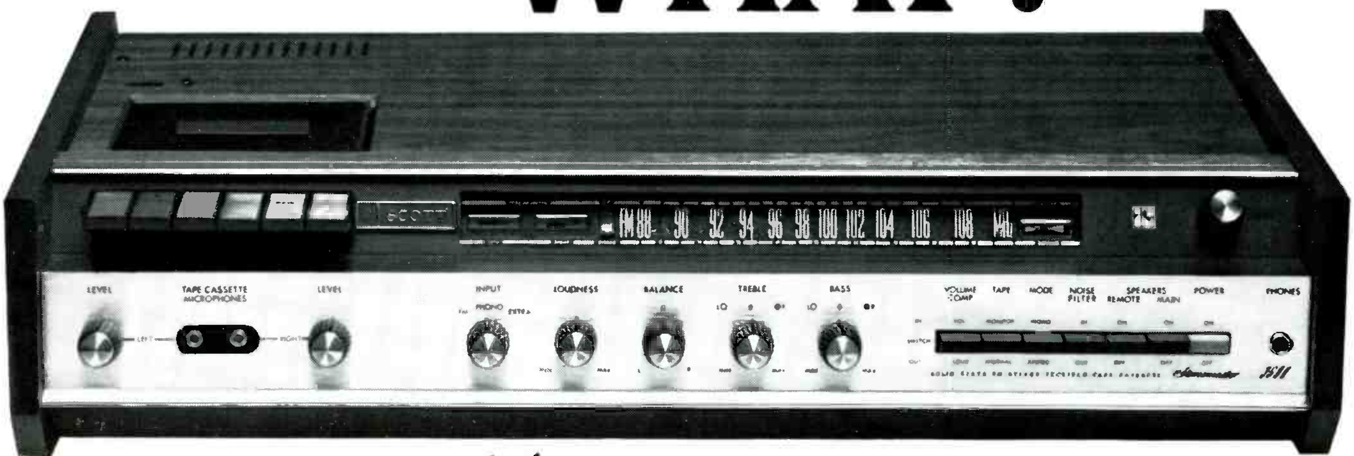

the Casseiver!®

IT'S AN FM STEREO RECEIVER!
IT'S A STEREO CASSETTE RECORDER!
IT'S THE NEWEST, MOST VERSATILE COMPONENT ON THE MARKET!

What's a Casseiver? Just a quicker way of saying Cassette/Receiver. Scott's new 3600 is an ultra-sensitive 65-Watt FM stereo receiver. It's also a professional cassette recorder with digital counter and individual record and playback meters. And it's all in one beautiful long low cabinet.

The Casseiver is versatile. You can listen to FM or FM stereo. You can listen to pre-recorded cassettes. You can also record onto cassettes, either from voice or instruments (there are two microphone jacks on the front panel), or from records (just connect a turntable), or directly from the Casseiver's own superlative FM stereo tuner. More? Add extra speakers or headphones. You can do a lot more with the Casseiver because it's a lot more than a receiver.

Inside, the Casseiver has a lot going for it. Scott's silver-plated FET front end brings in a raft of stations loud and clear . . . whether you live in the canyons of Manhattan or the Grand Canyon. Integrated Circuits, both in the IF strip and in the preamplifier, keep your favorite sound distortion-free and clear of annoying interference. The cassette section is specially built to Scott's demanding specifications, including a precision synchronous AC motor, assuring you of absolutely constant speed, with no annoying flutter or wow. AC operation is inherently stable, and requires no additionally stabilized power supply.

That's the Casseiver . . . a great new idea from Scott . . . An idea you'll get used to very quickly once you've seen and heard it in action. At your Scott dealer's showroom . . . only $399.95*.

3600 Casseiver Controls: (Receiver section) Inertia drive tuning control; Power on/off; Switching for Main, Remote, or both sets of speakers; Noise Filter; Mono/Stereo switch; Tape monitor control; Volume compensation control; Dual Bass and Treble controls; Balance control; Loudness control; Input selector; Center Tuning meter; and stereo headphone output; Balance Right, Balance Left; Tape selector, external or cassette. (Recorder section) Left and Right level controls; Dual microphone inputs; Left and Right Record level meters; Resettable digital counter; Individual controls for opening the cassette section, record, play, fast forward, rewind, and stop.

Specifications: IHF Music Power @ 4 Ohms, 65 Watts; IHF Music Power @ 8 Ohms, 48 Watts; Frequency Response ±1 dB, 20-20,000 Hz; Hum and noise, phono, −55 dB; Cross Modulation Rejection, 80 dB; Usable sensitivity, 2.5 μV; Selectivity, 56 dB; Tuner Stereo Separation, 30 dB; FM IF Limiting Stages, 9; Capture Ratio, 2.5 dB; Signal to Noise Ratio, 60 dB; Phono Sensitivity, 4mV.

SCOTT®

For complete details on the new 3600 Casseiver, write: H.H. Scott, Inc., Dept. 35-12, Maynard, Mass. 01754. Export: Scott International, Maynard, Mass. 01754.

*Recommended Audiophile Net.

Check No. 100 on Reader Service Card

December 1968 Vol. 52, No. 12

AUDIO

Successor to RADIO, Est. 1917

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Coming in January 1969

SPECIAL TAPE RECORDER ISSUE

Roundup of Tape Recorders—Specifications and details of the latest audio and video tape recorders.

How to Interpret Tape Recorder Specifications—Here's an Audio feature article that many readers have been waiting for: the meaning of each important tape recorder specification, with its relation to actual performance.

Auto Stereo Tape Players—Bert Whyte examines 8-track tape players, offering tips on installation and cartridge servicing.

The "Cross-Field" Tape-Head Technique—Fritjof Brodtkorb discusses the method and effect of using an extra tape head to introduce a bias field in opposition to the signal field near a recording head's gap.

. . . and other feature articles.

ALSO:

Equipment reviews, LP record and recorded tape reviews, ABZ's of FM, Audioclinic, Tape Guide, and other regular monthly departments.

ABOUT THE COVER: Books on hi-fi and electronics, stacked in a Yuletide package, depict an important source of information for audio enthusiasts and hobbyists. See reviews on page 18.

Audioclinic

JOSEPH GIOVANELLI

If you have a problem or question on audio, write to Mr. Joseph Giovanelli at AUDIO, 134 North Thirteenth Street, Philadelphia, Pa. 19107. All letters are answered. Please enclose a stamped, self-addressed envelope.

Amplifier Clipping

Q. Can you explain to me how an amplifier can deliver power beyond its clipping point?

It has been my impression that output increases as the input voltage increases, until clipping is reached. At this point no further increase in output can take place. Can you explain this. — Leonard Drasin, Louisville, Ky.

A. "Clipping" does not necessarily mean saturation of an amplifier. When an amplifier starts to clip, we do not know which stage is overloaded first. We hope that it is the output stage, but it can and does happen that the voltage amplifier will produce asymmetrical output before the output stage does this. Further, the output stage may also produce asymmetrical output. If this happens, you can see that one half of the stage will be producing relatively clean output signals. The portion of the output which is not clipping may provide more power output than the complementary element which is clipping, given a further increase in input signal beyond the point at which clipping is first observed.

This situation can usually be improved by replacement of the element which is clipping or by a rebalancing of the stage.

Remember, clipping does not mean that at a certain point all waveforms are flat-topped. If this was the case, no further power output could be obtained. However, all that is happening when a stage begins to clip is that its linearity is suffering. A given change of input signal no longer produces a proportionate change in output.

Of course, a properly designed amplifier will not clip in an asymmetrical manner. Further, it will be designed in such a way that the voltage amplifier will have considerable reserve capacity over that which is required to cause the output stage to begin clipping.

Even when the clipping action is symmetrical, this does not mean that the top of the wave has become completely flattened. It simply means that the positive and negative peaks of the wave are reduced in amplitude from the levels they would have attained, assuming that amplifier clipping had not taken place.

Therefore, while it is true that linearity suffers as the amplifier begins to clip, the total amount of power provided by this amplifier can be increased beyond the point where clipping action begins. We would hope that clipping can take place as far up on the power output curve of the amplifier as possible.

We would all like to dream of the amplifier that delivered its power in a straight, linear manner right to the point of absolute flat-topping. However, I rather doubt that this can be brought about. It would appear to me that transistor output stages would be more likely candidates for this kind of operation than would be true of tube devices. This is because of their tendency to be "turned on" fully at saturation. A transistor is almost out of the circuit in terms of having a very low internal resistance. This is not true of a tube which is completely saturated.

Midrange Speaker Damage

Q. Recently I discovered that my midrange speakers were inoperative. My wife reported that they had simply stopped functioning, and that she had not heard any unusual sound at that time. With a flashlight battery I determined that the voice coil was probably burned out. I am having them repaired. Because both speakers (one in each channel) were damaged, I suspect that perhaps a switching transient, with the volume on the preamp turned up, may have caused the damage.

While the speakers were being repaired, I temporarily replaced them with two, 3.2-ohm speakers from a TV set. Today I discovered that one of these speakers was completely burned out and that the other had very low output.

I am now worried that there may be a serious defect somewhere in my equipment. The amplifiers would appear to be at fault, but, because the damage occurs in both channels, and because I have two, separate mono amplifiers, it would mean that the same problem exists in both amplifiers. Is it possible,

then, that it resides in the stereo preamplifier, perhaps some kind of supersonic oscillation?—Max Prola, Jackson Heights, New York.

A. In solving this kind of problem, we have to think in terms of several factors.

First, perhaps the volume you are using causes more power to be delivered to your midrange speakers than can be safely handled by them.

Second, perhaps there are transient clicks which are causing your difficulty, especially when switching to the "phonograph" position of your preamplifier. The sound you hear when switching to this function might not seem loud to the ear, but the peak energy level might be enough to damage your midrange speakers.

Third, there is the definite possibility of oscillation somewhere in the chain of events between speaker and amplifier. This oscillation, however, should affect the tweeters before affecting the midrange speakers. This is apparent when you remember that the frequency of oscillation must be quite high or you will hear and recognize the fact that the equipment is oscillating. The tweeters are set up to receive high-frequency energy and should be damaged first by such oscillations. Of course, if you have a crossover network design wherein the highs are fed into the midrange speaker unattenuated, there is the possibility that the midrange speaker would be damaged. Even here, however, the tweeters should be damaged first.

Fourth, it is possible that your crossover networks are not operating correctly, allowing too much low-frequency energy to enter your midrange speakers. This surely will result in damage to many units. Midrange speakers are not so rugged as woofers because their power requirements are not so stringent as those for a woofer.

Let's assume for the sake of argument that the trouble really is in the form of some kind of supersonic oscillation. I doubt, frankly, that this is the case, but here is a possible method by which you can localize the source of such oscillations:

See if you can observe them at the amplifier's output terminals, using a VTVM or a 'scope as the indicating instrument. Do not feed a signal into the equipment when making your initial test. After the initial checks, feed a signal into the amplifier. Does the voltage appearing across the amplifier's output terminals vary with modulation as one would expect? Does it take sudden, unexplained jumps, having little to do with modulation level? If the latter is true, you have oscillation which is dependent upon the level of input signal.

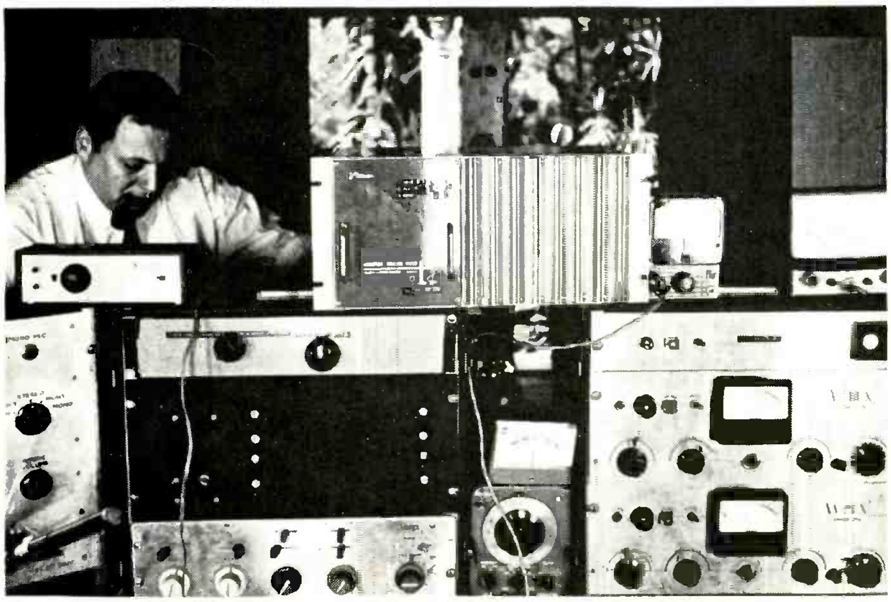
It does seem odd that this oscillation could take place in two separate amplifiers, but it is possible. However, if such oscillation is taking place, it is more than likely the result of either a fault in your preamplifier or a fault in the manner by which your cabling was done. What I mean by this is that your input and your output leads may be close together and running parallel to each other. Check this at all sources—cartridge leads, tuner cables, power amplifier, amplifier input, etc. All of these leads should be shielded, kept away from one another and from the speaker lines.

You may find that the oscillation, if present at all, occurs with certain settings of the volume control. If it does, this tends to confirm my diagnosis.

Transient clicks can be suppressed by connecting five-megohm resistors across outputs of such items as tuners, tape recorders, preamplifiers, the phono stage within a preamplifier, etc.

The great hall of the Hammond Museum. This room is the location of the organ played by Richard Elsasser on Nonesuch H-71200 ("Yankee Organ Music") and H-71210 (Organ Symphony No 5 by Charles-Marie Widor)

AR-3*a* speaker systems were designed for home music reproduction. Nonesuch Records uses them as monitors at recording sessions.



Nonesuch Records recently recorded several volumes of organ music played by Richard Elsasser at the historic Hammond Museum near Gloucester, Massachusetts. To make the recording, Marc Aubort of Elite Recordings, engineering and musical supervisor, used Schoeps microphones, and Ampex 351 recorder, Dolby A301 Audio Noise Reduction apparatus, and several pieces of equipment which were custom made. To monitor the input signal and to play back the master tape, Aubort used an AR amplifier and 2 AR-3a speaker systems.

The AR-3a speaker system is priced from $225 to $250, depending on finish.

ACOUSTIC RESEARCH, INC 24 Thorndike Street, Cambridge, Mass 02141
Overseas Inquiries: Write to **AR International** at above address.

Check No. 5 on Reader Service Card

What's New In Audio

Panasonic Enters Component Field

Panasonic has introduced four FM stereo receivers, in addition to three new reel-to-reel tape recorders and several speaker systems, to mark its entry into the stereo hi-fi component field. The top receiver, Model SA-4000, is a 160-watt FM stereo unit that features automatic motor tuning (as well as manual motor tuning), a five-station

pre-selection pushbutton system, four FETs, and six ICs.

Among other features are bass and treble "switches," input level controls, three-position muting selector, and a black-out dial glass with "LuminaBand Tuning" instead of dial pointers. Price is on request, though the next-in-line receiver, a 90 watter, priced at $349.95, can serve as a guidepost.

Check No. 6 on Reader Service Card

Sansui 70-Watt AM/FM Stereo Receiver

Sansui's moderately priced Model 800 AM/FM stereo receiver ($259.95) features 70 watts total IHF power, and an FET front end. The solid-state unit has a four-way front-panel speaker selector switch, stereo headphone jack, friction-coupled tone controls, high-frequency filter, loudness control, an illuminated tuning meter, and a tape monitor switch. The FM stereo section

incorporates automatic stereo switching circuitry, while the input antenna terminals are designed for both 75-ohm and 300-ohm antennas.

Check No. 10 on Reader Service Card

Cassette Tape Recorders

The number of cassette tape recorders on the market continues to increase, giving consumers a wider-than-ever choice. Sony Superscope, for example, recently introduced its new TC-124 "stereo cassette-corder." Measuring six by nine inches, the unit incorporates a pop-up lid and cassette ejector for handling of both 60- and 90-minute tape cassettes. Recording level in the stereo record mode is adjusted automatically. A meter indicates the condition of batteries, which have a 5½-hour play life (can be extended to 7 hours per charge with an optional ($14.95) Sony BP-16 nickel cadmium battery pack). A cardioid stereo microphone with start/stop switch, C-60 tape cassette, personal earphone, two patch cords, and four batteries are included. Specifications include a wow and flutter of 0.28% and a signal-to-noise ratio

of 45 dB or better. Priced under $169.50. A Model 124CS is also available, with two full-range extension speakers and a black vinyl tote bag, for less than $199.50.

Check No. 12 on Reader Service Card

Ampex adds the Micro 90 cassette playback deck to its line of recorders. It has an automatic changer that plays one side of up to six cassettes automatically. Pushbutton controls include fast forward, rewind, stop, pause, play/reject, and on/off. Dimensions are 15¼-in. W x 9¼-in. D x 4⅜-in. H (with cassette automatic sleeve mounted,

6¾-in.). Priced at $129.00. A record/playback version, including two matched 6-in. x 9-in. speaker systems in a walnut case and a 20-watt (peak) stereo amplifier; two dynamic, omnidirectional microphones with detachable stands, and a record level meter, is priced at $269.00.

Check No. 14 on Reader Service Card

New Scott 3-FET Tuner

H. H. Scott's new FM stereo tuner, Model 312D, features three field-effect transistors in the front end and an integrated circuit i.f. strip.

Usable FM sensitivity is 1.7 μV; capture ratio, 1.9 dB; crossmodulation rejection, 90 dB; selectivity, 46 dB; stereo separation, 40 dB.

A front-panel meter switch permits the meter circuit to be used to indicate signal strength, zero-center tuning, or multipath indication. There's an oscilloscope output, too, for checking multipath distortion. Other front-panel provisions include: controls to vary the

level of both headphone and amplifier outputs independently, an interstation muting control, pushbutton switches, front-panel output for direct tape recording. The unit also has automatic switching in the presence of a stereo signal. $319.95.

Check No. 16 on Reader Service Card

Benjamin's Elac Cartridges

Benjamin Electronic Sound has a new series of Elac moving-magnet stereo cartridges. Consisting of three basic models, the top-of-the-line elliptical-stylus-equipped 444-E, at $69.50, is said to have a virtually flat frequency response from 10 to 24,000 Hz, channel

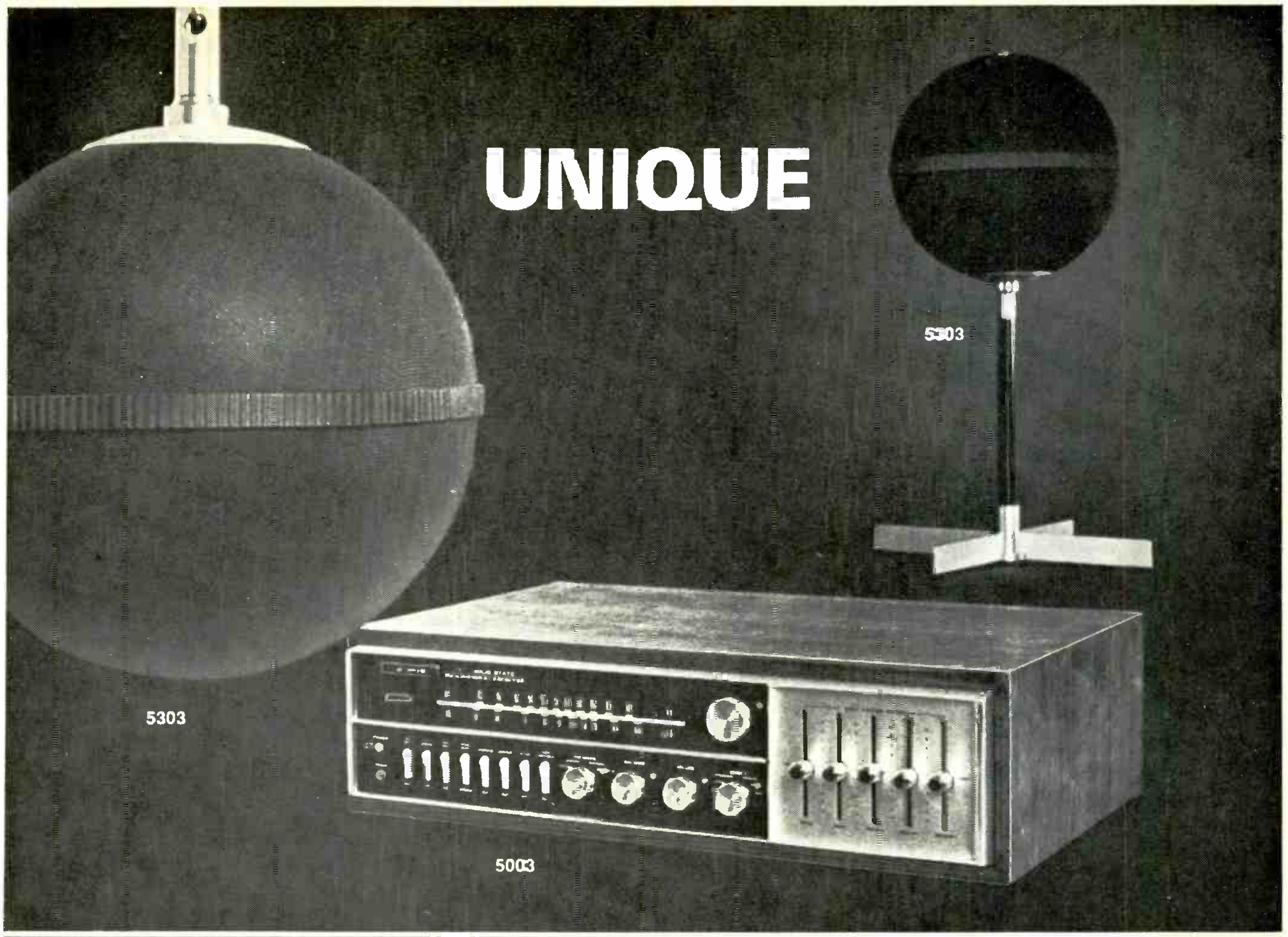
separation better than 26 dB at 1000 Hz, and operates with a tracking force as low as ¾ gram. (A model 444-12 has a 0.5-mil diamond stylus instead of an elliptical-shaped stylus, and it's priced at $59.50.)

The Elac model 344-E, with elliptical stylus at $49.50, covers the frequency range from 20 to 22,000 Hz. virtually flat, according to the company Optimum tracking force is from one to two grams. (A model 344-17, priced at $39.50, is equipped with a 0.7-mil diamond stylus.) Model 244-17, priced at $24.95, offers a frequency response from 20 to 20,000 Hz. with a recommended tracking force of 1.5 to 3 grams. The 244-17 is equipped with a 0.7-mil diamond stylus.

Check No. 18 on Reader Service Card

New Record Cleaner

A new Watts hi-fi "Parastat," intended for use on new records or records in new condition, has been announced by Elpa Marketing Industries. The record-cleaning and maintenance kit consists of a brush with nylon bristles that, when flexed slightly, pro-



Shown above and described below are just a few examples of the most unique and formidable line of stereo equipment in the world today. From powerful stereo systems, to all-in-one compacts, to breathtaking individual components, there is a model designed for everyone from the most ardent stereo enthusiast to the casual listener.

Model 5303—Powerful Spectrum Speaker System—Non-Directional Sound Total sound diffusion—a full 360 degrees. Four free edge woofers and four horn-type tweeters in hermetically sealed metal enclosures to handle up to 80 watts in input. Frequency response range from 20 to 20,000 Hz. May be pedestal-mounted or suspended from the ceiling. *Diameter: 13½" 26.4 lbs.*

Model 5003—140 Watt Solid State AM/FM-FM Stereo Receiver with exclusive "Sound Effect Amplifier" Tone Control System
Full 140 watts power output. All solid state FET circuitry with five IF stages. Automatic stereo switching, two speaker system selector, stereo and fine tuning indicators. Full complement of inputs, jacks and terminals with matching controls. *5⅞" H, 20¾" W, 14¼" D 30.8 lbs. w/cabinet*

Model 1684—Solid State 4-Track Stereo Tape Deck—Built-in pre-amplifier for superb reproduction at 7½ and 3¾ ips. 7-inch reels. Automatic stop device, professional VU meters, 3-digit tape counter, DIN and pin jack connectors. Accessories include full and empty 7-inch reels, DIN cord, splicing tape, dust cover and two reel clamps. Oil-finished wooden cabinet.
12 transistors 15⅝" H, 13⅝" W, 6¾" D 22 lbs.

Model 6102—Deluxe Automatic 4-Speed Stereo Turntable and 8-Track Stereo Player—Large 11-inch platter for wow and flutter characteristics less than 0.3%. Tubular tonearm with moving magnetic cartridge and diamond stylus. 8-Track Stereo player features a 6 transistor preamplifier and wow and flutter characteristics of less than 0.3%. Fine furniture finished wood with molded acrylic dust cover.
9½" H, 17¼" W, 13¾" D 23.4 lbs.

Model 5001—60 Watt Solid State AM/FM-FM Stereo Receiver with exclusive Sound Effect Amplifier—With built-in Sound Effect Amplifier (SEA), you have complete freedom and control of the total tonal spectrum in five different frequencies: 60, 250, 1,000, 5,000 and 15,000 Hz. New FET circuitry with four IF stages. Automatic stereo switching. *5⅞" H, 20¾" W, 14¼" D 30.8 lbs. w/cabinet*

JVC

Manufactured by Victor Company of Japan, Ltd.

For additional information and a copy of our new 28 pg. full color catalog, write: Dept. J,

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Check No. 7 on Reader Service Card

duces a pressure from each bristle of about $\frac{1}{10}$ of a gram on the surface of a disc. According to Elpa, bristles thoroughly explore each groove, right down to the bottom, so that every trace of foreign matter is removed.

Two vials are included with the kit: one with distilled water (which can be replenished at a local drug store), the other with an anti-static fluid. Both fluids are applied to a pad which is kept in the brush case. The brush is wiped against the moist pad before it is used on the playing surface of a record. $15.00.

Check No. 20 on Reader Service Card

Pioneer's High-Efficiency Speaker

Pioneer Electronics' new CS-5, an intermediate-size bookshelf speaker system, is an air-suspension type that is said to exhibit high efficiency. Thus the manufacturer claims it is suitable for low-power-output amplifiers.

The two-way speaker woofer uses a specially developed cloth surround for greater compliance and a long-throw voice coil; a cone type of tweeter is employed. The enclosure is an oiled walnut cabinet, with a pebble-weave grille cloth that is removable. The enclosure's back is fitted with a wall hanger in the event that a user chooses this installation format. Price is $59.00.

Check No. 22 on Reader Service Card

New E-V Console Speaker System

Electro-Voice has introduced a new console speaker system, the E-V Six-B. It incorporates two 12-in. woofers that employ a new plastic foam cone suspension which, according to the manufacturer, eliminates boominess from bass tones. The system also includes an 8-in. mid-range speaker and a tweeter with an aluminum voice coil. The 8-ohm system can handle 60 watts of program material. Speakers are encased in a picture-frame cabinet, while a pedestal base "floats" the cabinet. In addition, the company advises that the grille cloth creates an unusual effect, with a change in color and texture occurring when one's position is changed or the light source changes. The console, which measures 25-in. H x 20-in. W x 15-in. D, is priced at $299.95.

Check No. 24 on Reader Service Card

Letters

Color TV Audio Jack

• Mr. J. Pushkin reports reading in your column [*Audioclinic*] the non-existence of a color TV with audio detector take-off.

We are pleased to report that the Andrea 22″ Imperial color TV component chassis incorporates an audio output jack which permits connecting the recovered audio at full fidelity directly from the sound to an existing and separate Hi-Fi system.

Raymond Masvidal
Andrea Radio Corp.
Long Island City, N. Y.

• *Thanks. Specifications of the Andrea "Imperial," however, indicate that the audio jack's output impedance is 50,000 ohms, which limits length of connecting cable to a hi-fi system. A cathode follower would be more desirable since its low-impedance output would permit long runs of connecting cable without deteriorating sound quality.—Ed.*

Melodiya's Recording Blanks

• Perhaps I can clear up one of the mysteries surrounding Dr. Ray Dolby's visit to the Moscow studios of Melodiya Records, reported by Bert Whyte in his "Behind the Scenes" article in your October issue. "I found they were using American Transco lacquers," Dr. Dolby is quoted as saying. "I don't know where they obtained the Transco lacquers and they didn't say."

Clearing up the first mystery is easy: Melodiya has been buying Transco recording blanks from us for many years. As to the second mystery, why "they didn't say," perhaps it is just a case of "does Macys tell Gimbels?" Could it be that a home grown Columbiya or RCA Vanya is looking for that official Russian state agency?

Max Roth
Transco Products International
New York, N. Y.

An Octave to the Right, Please

• I am writing to you about an error that appears in Audio, Sept., 1968 [Electronic Organs] on the chart at top of page 28. The second (brown) drawbar from the left is pitched at $5\frac{1}{3}'$ in organ pitch, and not at $10\frac{2}{3}'$, as this chart shows. It is, of course, out of sequence, as the first (brown) drawbar comes in at sub-unison pitch, but the second drawbar really sounds between the second and third drawbars, and not between the first and second drawbars, as your chart implies. The drawbars are pitched, from left to right, as: $16'$ $5\frac{1}{3}'$ $8'$ $4'$ $2\frac{2}{3}'$ $2'$ $1\frac{3}{5}'$ $1\frac{1}{3}'$ $1'$.

Therefore you should have placed the second vertical line in the space where it says "note played" instead of where it is.

May I suggest also that many Baroque organs have tremulants that change loudness and therefore timbre, but very little pitch in their pulses. High-wind-pressure organs (Romantic and theatre) have tremulants that change pitch markedly, and, naturally, also loudness and timbre too; all three markedly.

Stevens Irwin
St. Petersburg, Fla.

For our president's birthday, we baked him a cake.

And gave him the first production model of our new Project "M".

Haskel Blair celebrated his birthday just as we turned out the first Project "M". The timing was so perfect, we gave it to him for his birthday.

But why give a speaker system to the president of a firm that makes so many speaker systems?

You see, we all believe the new Project "M" is our finest achievement in compact speaker systems. (And we've been building fine speakers for over 30 years.)

Besides, we certainly put a lot of work into it. When we started on the Project "M", we wanted to develop an entirely new and superior speaker system that would surpass the lows and highs of any high-quality, high-priced compact bookshelf system.

To achieve that we began with some pretty sophisticated design principles and ended up creating a new woofer and a new tweeter and even a new enclosure. But it was worth the effort. The result is the remarkable Project "M".

To give you an idea of how good the Project "M" really is, we took our first two production models and conducted a blindfold listening test with Mr. Haskel Blair matching the Project "M" against the two speaker systems he has in his home. He thought the Project "M" sound was so great, we gave him the two units for his birthday. Oh yes, and he was very surprised and pleased when we told him the project "M" would be selling for $99.50!

How can the Project "M" do it? Well, for one thing, Project "M's" all-new woofer has our secret new surround material which permits the cone to make ½" excursions with a linearity that is accurate to within 0.1%. This means that our woofer has a linearity that is better than the minimum distortion capability of even the best amplifiers.

Together with an equally new tweeter, the system puts out a magnificent sound along the entire audible range, with an exceptionally smooth frequency response from below 30 Hz to beyond 20,000 Hz.

Project "M" is housed in an attractive 23½" x 12¾" x 11⅞" oiled walnut enclosure that is finished on four sides for vertical or horizontal placement.

Compare the sound of our Project "M" with any other compact speaker system, the more expensive the better. Forget the price until you've compared the sound. After that, compare the price. Then you'll know why we gave our president the first Project "M"!

UNIVERSITY® SOUND
A DIVISION OF LTV LING ALTEC, INC.
9500 West Reno • Oklahoma City, Oklahoma 73126

Listen . . . University sounds better!

Check No. 9 on Reader Service Card

BEHIND THE SCENES

BERT WHYTE

WALK INTO the average hi-fi shop and what do you see? Walnut boxes. Dozens of them... most quite small, resembling breadboxes or whiskey cases. Some are even as big as *two* whiskey cases. These are loudspeakers... a ubiquitous breed, many imitative of each other's design. The claims for these speakers range from modest to monumental. No matter how clever their design, however, to *me* they have the common failing of sounding like a small point source. The music they reproduce sounds rather compressed and constricted, like it is being squeezed through a small window.

Am I prejudiced? You bet! For you see, I'm a "big speaker" man. Unabashed. Uncompromising.

You'll see the big ones in some of the better salons, interspersed among the walnut boxes, standing like some noble monoliths; speakers of massive proportion and imposing aspect. These are the sonic aristocrats . . . the "big speakers" . . . loathed by women and loved by men.

At the moment, this partiality is a bit of a problem since I find myself on a most uncomfortable spot. You see, I've been listening to a pair of small speakers that are reproducing music with a high-quality BIG speaker sound, something no small speakers have any right to do. This iconoclastic speaker is known as the Bose 901 . . . and thereby hangs a tale.

The 12-Year Quest

The speaker was designed by Dr. Amar G. Bose, a Professor of Electrical Engineering at the Massachusetts Institute of Technology. Among other things, Dr. Bose teaches acoustics, and for many years has been an avid music lover and hi-fi enthusiast. About 12 years ago, he began a research program to investigate loudspeakers, realizing the truth of the old axiom that the loudspeaker is the weakest link in the chain of music reproduction. A student of the violin in his younger days and a frequent visitor to concerts of the Boston Symphony Orchestra in nearby Symphony Hall, Dr. Bose could not equate what he heard in the concert hall with the reproduction of music through commercially available loudspeakers. This was especially true with his beloved strings, which the speakers reproduced as "interesting but far from realistic sounds." Preliminary investigations convinced him that a good part of the loudspeaker problems derived from the inadequacy of standard speaker measurement techniques, which relied mainly on anechoic chamber tests or free-field testing, and a lot of subjective listening.

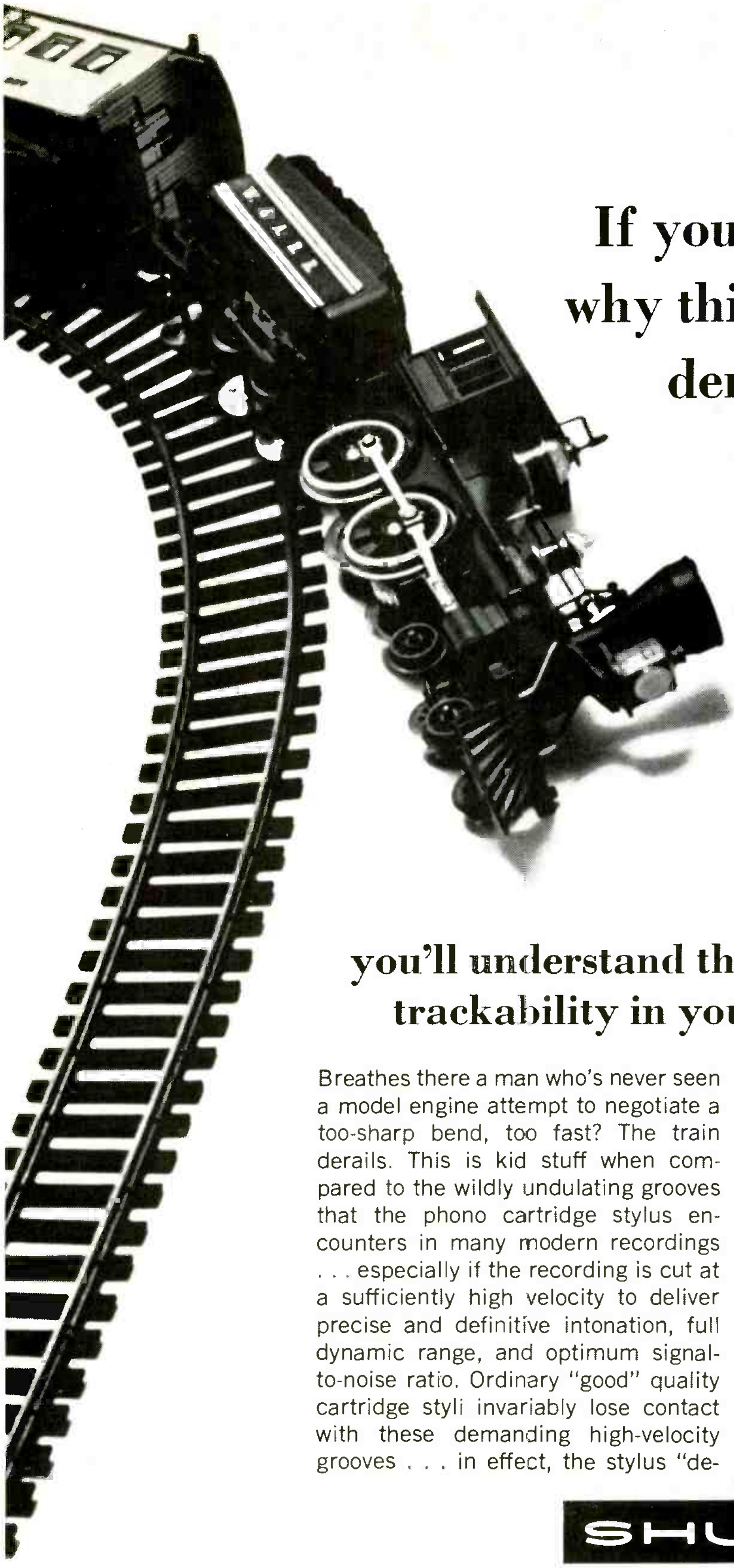
Speaker design over the past thirty years has been part science, with anechoic chambers and chain-driven oscillators and oscilloscopes, and part art, which is "cut and try," "cone doping," "individual tuning," and still more subjective listening. There would seem to be little argument that empiricism plays a significant role in speaker development. It is also obvious that we don't listen to music in anechoic chambers, so that questions have been raised regarding the relevancy of chamber data to the home listening environment.

The task Dr. Bose set for himself was formidable indeed: to determine what kind of sound an ideally "perfect" speaker (theoretically a pulsating sphere) would make—if such a thing could be made; and to develop precise techniques of measuring this sound, all within the context of the home listening experience.

With the vast technical resources of MIT at hand, Dr. Bose's first consideration was the simulation of the ideal pulsating sphere, whose surface will emit pressure waves in all directions evenly over its entire circumference. It is important to note that a prime requirement was that the pulsating sphere be measured in a typical room, in the same fashion as a normally operating speaker. Thus the normal room nodes, standing waves, etc., would be included in the measurement. The ideal pulsating sphere turned out to be a high-voltage spark, which was discharged into a typical home-type room at a frequency rate of once per second. The sound of the spark discharge was picked up by a calibrated Western Electric 640-AA condenser microphone. The microphone reconverted the sound into an electrical signal

Fig. 1—A front and back view of the Bose 901 speaker system, with grille cloths removed, is shown below. (Note that the rear of the system has eight speakers, while the front panel has only one.) At right is the system's active equalizer, which has 10 transistors.

Front Back



If you understand why this model train derailed . . .

you'll understand the importance of high trackability in your phono cartridge

Breathes there a man who's never seen a model engine attempt to negotiate a too-sharp bend, too fast? The train derails. This is kid stuff when compared to the wildly undulating grooves that the phono cartridge stylus encounters in many modern recordings . . . especially if the recording is cut at a sufficiently high velocity to deliver precise and definitive intonation, full dynamic range, and optimum signal-to-noise ratio. Ordinary "good" quality cartridge styli invariably lose contact with these demanding high-velocity grooves . . . in effect, the stylus "derails". Increasing tracking weight to force the stylus to stay in the grooves will ruin the record. Only the Super Trackability Shure V-15 Type II Super-Track® cartridge will consistently and effectively track all the grooves in today's records at record-saving, less-than-one-gram force . . . even the cymbals, drums, orchestral bells, maracas and other difficult-to-track instruments. It will make all of your records, old and new, sound better. Independent experts who've tested the Super-Track agree.

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which was fed into an accumulator in a special digital computer.

When 4000 discharges had been stored, the computer performed a computation known as digital convolution or superposition integration (which I assure you is quite beyond my ken) and the result was, in essence, a mathematical model of the ideal speaker. The signals on the computer tape are, of course, different from the original (reproduced by the ideal pulsating sphere/spark speaker) because of the characteristics of the "speaker" and the acoustical properties of the room. However, there is a mathematical relationship between the two signals. Once this is known it is possible to calculate how the "speaker" would reproduce any other recorded signals in that room, even something as complex as a symphony. The next step was to feed actual speech and music samples into the computer in the form of electrical signals. The computer produced a tape of this material as it would have been recorded in a living room if the samples were played through a hypothetical one-eighth of an ideal pulsating-sphere speaker placed in a corner. The same music and speech samples were also played in the same room through an approximation to the ideal sphere speaker, consisting of 22 four-inch speakers placed on an octant of a sphere of 20 inches radius, driven by a computer-derived electronic equalizing network.

The two tapes were subjectively compared. According to Dr. Bose, observers were unable (during an A-B test) to distinguish the computer-processed music and speech from the speaker-processed music and speech.

The mind boggles at the enormous computations necessary to produce these tapes. The need for a computer is obvious. Even so, when Dr. Bose started this experiment some years ago, it would have taken 20 *hours* of computer time to process 7 *seconds* of music! Today, this has been reduced to 3 to 6 minutes for the same 7-second processing.

The importance of the experiment was that it proved that with proper frequency equalization, the multiplicity of closely spaced small speakers on the spherical surface can produce music and speech signals in a *normal listening environment* that are *subjectively* indistinguishable from those that would be produced by an ideal pulsating sphere in the same environment having no resonances, phase shift, diffraction, or distortion of any kind. Thus Dr. Bose had a precise measurement technique, yet one that worked with the subjective factors in listening experience.

Another experiment was conducted to provide a measurement technique for speaker distortion. Distortion in speakers has usually been measured in an anechoic chamber, but this method of detecting distortion isn't meaningful to the listener in the environment in which the speaker normally operates—his living room. Most speakers exhibit various forms of distortion as their intensity of radiation increases, and have relatively negligible distortion at very small intensities. Dr. Bose's experiment made use of this fact to determine the level of intensity at which a speaker begins to generate audible distortion in music reproduction. Selections of music and speech were played a number of times through the speaker in a listening room at successively increasing intensity levels. Binaural recordings of each successive speaker playing were made at a constant, standard level. The various recordings were synchronized on parallel tracks of an eight-channel tape recorder. The listener subject was then given an A-B presentation of the sample that was recorded with the speaker playing at the lowest level and a sample representing a higher speaker output level. Naturally, both samples were presented at the same level to the listener, and he was asked only to try to detect a *difference*. The level of intensity produced by the loudspeaker for which a difference is first detected is then a measure of the performance of the loudspeaker. This measure is pertinent to the ultimate *subjective* evaluation, but it was obtained without introducing the problems of individual value judgment or prior notions of the sound of distortion. Essentially, this is a way of correlating objective measure with the subjective perception of sound. The subject was not asked whether he likes or dislikes a sound, but merely if he could detect a *difference* between sounds.

Armed with these unique measurement techniques, Dr. Bose continued his researches. As you can see from the foregoing, there has been considerable emphasis placed on the measurements as applied to the listening room environment. There have been many studies of concert halls made which show, conclusively, that an auditor at a concert, no matter where he may be seated, hears the orchestra at a certain ratio of direct-to-reflected sound. Dr. Bose expanded much of the original European data on concert hall measurement; his studies have shown that for virtually all seats in the average concert hall, the reverberant field is dominant. Even in a large hall such as Symphony Hall in Boston, the reverberant field equals the direct field only 19 ft. from the orchestra. It is important to note that in a reverberant field, sound energy arrives at any point via reflections from the surfaces of the room and that the angles of incidence of the arriving sound are widely distributed. This spatial property of the sound incident upon a listener, plus the frequency spectrum for the incident energy, are the important factors in the subjective appreciation of music.

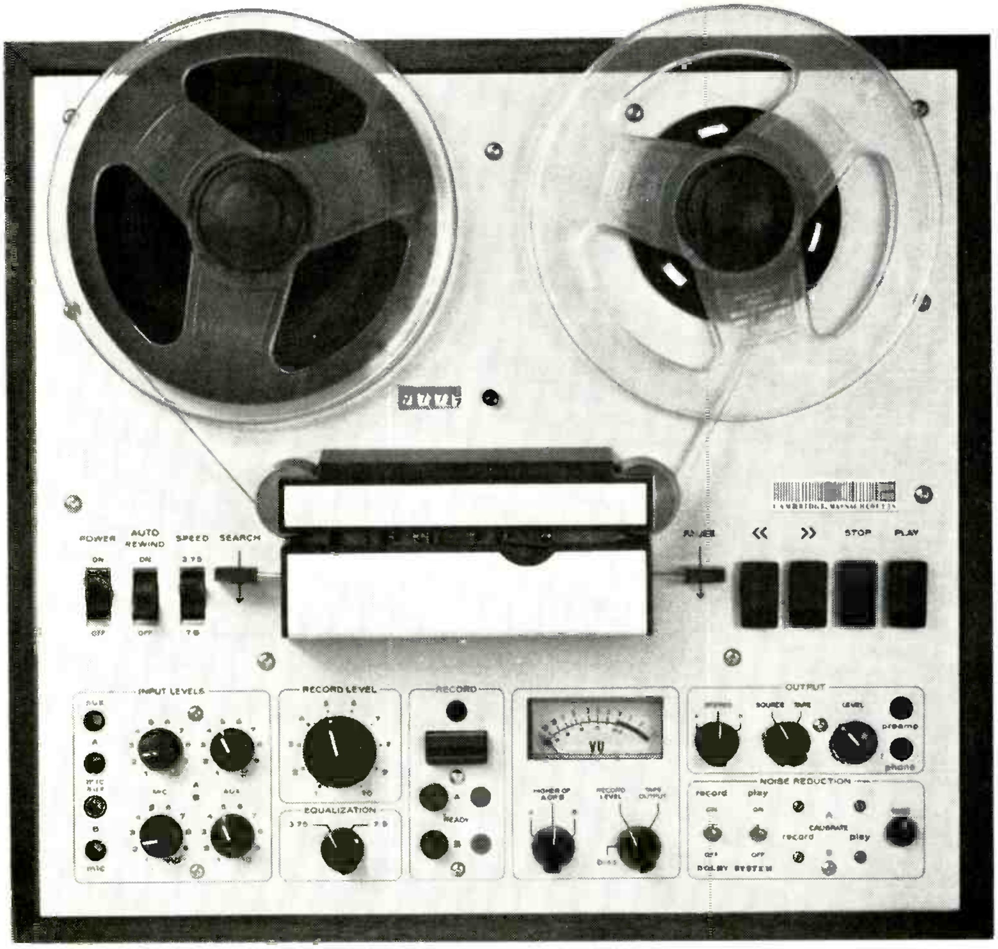
* * *

At first glance the Bose 901 is rather unprepossessing, but a closer look reveals that the unit is pentagonal in shape, with the two rear panels forming a "Vee" angled at about 30 degrees. As you can see from the illustration, four speakers are mounted on each of the rear panels (total of eight), while a single speaker is mounted on the front panel. All nine drivers are identical, specially designed, 4-in. units, with long-excursion voice coils. Each speaker has a power handling rating of 30 watts. Each is a full-range speaker, so there are no crossover networks. The sealed enclosure is densely packed with fiberglass. In this seemingly simple unit there are many sophisticated concepts derived from the research program.

The Bose Model 901 is sold as a stereophonic speaker *system*, with two of the units comprising the stereo channel, and both units controlled by a specially matched active equalizer. To function properly, the speakers must be placed in front of a wall with a minimum spacing from the wall of 6 inches, a maximum of 18 inches (optimum spacing is 12 inches).

Dr. Bose's concept, based on the concert hall measurements which showed the need for a dominant reverberant field, results in the 8 speakers on the two back panels radiating 89 per cent of the energy toward the wall, while the single speaker on the front panel radiates 11 per cent of the energy into the room. This is said to provide the necessary ratio of reverberant-to-direct sound. With the rear panels angled at 30 deg. to the wall, the wavefronts emerging from the first reflections from the back wall, together with the second reflections from the side walls, produce an effective source that is much larger than the actual enclosure; and that is well distributed across the wall in the

The First KLH Tape Recorder



IN a recent demonstration, the $600 KLH home tape recorder was compared to a professional machine that sells for some $3,500.

Both recorded from the same wide-range, noise-free source; in fact all conditions of comparison were equal, with one exception: The KLH* Model Forty operated at 3¾ ips, quarter-track, the professional machine at 15 ips, half-track.

When the recordings were played back, listeners said the only difference they heard was $2,900.

KLH Research and Development Corp.
30 Cross Street, Cambridge, Mass. 02139

Gentlemen: Please continue. How can a 3¾ ips tape equal a 15 ips tape in sound quality? Why only one VU meter on a stereo recorder? What are those two little switches that say "Dolby System" underneath? etc.

Name____________________

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same sense that an orchestra is distributed across a stage. Since the reflected sound affords a virtual central image, considerable flexibility is possible in regard to the separation of the speakers. Naturally there is a certain optimum placement in any given room, but if circumstance dictates, the speakers may be placed further apart and closer together than other types of speakers, while maintaining the desired balance of stereo separation or center fill (judicious angling of the speakers inwards or outwards in relation to the walls is necessary, of course).

The nine speakers in each enclosure are full-range units and, as you can see in the photo, are closely spaced. The resultant acoustic coupling causes the resonant frequencies of each speaker to be different from every other speaker. In much the same manner as i.f. cans are staggered to get a broad response, the resonances become inaudible; thus the sound of the total array of speakers is very smooth. With no crossover network, each speaker receives the same audio signal. At low frequencies, the nine drivers, operating in phase, move a great deal of air, which is a well-known principle. At higher frequencies the relatively small cones behave like tweeters, with the highest frequencies propagated by the speakers' dustcaps. The pentagonal shape of the enclosure ensures there are no sides parallel to any of the panels on which any of the nine drivers are mounted, which eliminates the problem of standing-wave resonance.

The Bose active equalizer is a solid-state device which influences the acoustic output of the two speaker units. Precise equalization is necessary for flat frequency response of the radiated sound. This involves considerable boost. In the case of flat response for the lowest frequencies, as much as *18 dB* boost is needed. By manipulating the controls of the unit in the proper manner, a total of 20 different frequency contours are available to compensate for variations in room acoustics, phono pickups, program material, and so on. Some of the contours available are unique. For example, midrange frequencies can be reduced, but without a drastic roll-off of high frequencies. There is also a "below-40-hertz" switch that can reduce turntable rumble and other low-frequency disturbances without adversely affecting response above 50 hertz. The equalization of the signal comes before the power amplifier. The unit can be conveniently patched into a preamp/amplifier setup, or an integrated amplifier or a receiver. In fact, the unit is treated as if it were a tape machine, utilizing the tape-in/tape-out jacks of your particular rig. Auxiliary tape input and output jacks are provided on the rear panel of the equalizer.

* * *

Having described the Bose 901 system and its revolutionary concepts, the question, quite naturally, is how does it sound? At this point we'll have to bring up the matter of power amplification. [Because the equalization comes before the power amplifier] when you are trying to reproduce the very lowest frequencies at substantial levels, the 18-dB boost mentioned earlier can place really heavy demands on amplifier power. Let's put it this way: for 99 per cent of the program material played by most people at better than "apartment house" levels, a good 30 to 40 watts *continuous* power per channel should suffice. But when you want to play that other 1 per cent of program material — the great pipe organs, with the really low pedal and, most especially, a sustained low frequency at thunderous levels — that's when you need all the amplifier power you can afford.

The speakers can handle high power, of course (each speaker in the Bose enclosures is rated at 30 watts, for a total of 270 watts per channel). You can lower your power requirements by activating the "below 40 Hz" switch, however. You gain about 6 dB this way. It is true you lose the frequencies below 40 Hz, but if you desire to play the system at house-shaking levels, and don't have sufficient amplifier power, this is the way it can be accomplished.

I've had good results driving the Bose with medium- and high-power amplifiers. I tried it with a University receiver and with a CM 911, for example. Also with the McIntosh 2105, which has the advantage of a power level meter. If you monitor the output of the Mac amplifier during the heaviest fortes of some grandiose symphony, and set the gain so that you don't exceed +3 (105 watts per channel), you won't hear any distortion. (I didn't.) You will also find it's pretty loud!

Lucky lad that I am, there is very little program material that I can't cope with; not when I have a pair of McIntosh 3500 amplifiers, each capable of a mere 350 continuous watts per channel! These 120-pound (each) brutes are quite magnificent when working with the Bose speaker systems. I put on an organ recording containing plenty of "low C" pedal, and at a level of 108 dB (measured on a sound-level meter) the visual evidence of a scope across the speaker terminals and the audible evidence indicated that the Bose and the Mac had just barely reached a point of mutual distortion.

In evaluating the sound of the Bose 901 system, I was constantly aware that I was listening to an entirely *different* kind of loudspeaker. In some ways, comparison of this speaker with conventional types isn't quite valid. Even without its other virtues, the overwhelming superiority of the Bose in terms of *spatial presentation* and *stereo effect* was immediately apparent. There is no question in my mind about the desirability of the direct/reflecting principle for home listening. With a symphonic recording, the illusion of an orchestra spread across the wall is uncanny. With the virtual image in the center you can sit well away from the traditional "stereo axis" and enjoy a very good stereo presentation. It is the closest thing I have heard to a true three-channel stereo recording. The overall sound is outstanding for its clarity, transparency, wide range, crisp clean transients . . . and just plain naturalness. Equally astonishing is the bass response. To hear a thunderous "low C" organ pedal from these small (approx. 20 x 12 x 12) speakers, or a clean, weighty impact of a large bass drum is truly impressive. The speakers are mercilessly revealing of the faults in all manner of program material. It is a comparatively rare recording that could be played with the equalizer set in the flat position. (There is obviously more high-frequency distortion around than I realized.) The Bose really comes into its own with top-quality tapes. I have some symphonic masters and some one-to-one copies of masters, and they were reproduced with a naturalness that is quite compelling.

With only kudos for the Bose's performance, there *are* some minor drawbacks, some of which have already been mentioned: Fairly high-power amplifiers are needed to realize the 901's *full* potential; if you use headphones, the equalizer should be switched off; priced close to $500 for the entire system (two speaker systems and the solid-state equalizer), they cannot be considered to be truly inexpensive; placement requirements (6 to 18 inches from the wall) can be a problem to some persons since "bookshelf" location is excluded. But the above are trifles, in my estimation, when balanced against the astonishingly realistic sound achievable at home with the Bose speakers.

There is no doubt that the much-abused and overworked term, "breakthrough," applies to the Bose 901 system and its bold new concepts. Æ

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Tape Guide

HERMAN BURSTEIN

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FM Pickup

*Q. Now that my *** tape recorder is working satisfactorily (thanks at least in part to your guidance), I would appreciate your assistance with an interference problem. I am located less than a mile from an FM transmitter, and the recorder picks up its signal audibly whenever it's turned on, regardless whether the amplifier and tuner which comprise the remainder of the system (to which the recorder is connected) are energized. Fortunately, the interference is audible only in playback mode, but I would like to get rid of it.*—Stanley Metalitz, Kensington, Maryland

A. It seems that the FM signal is being picked up and rectified by the first playback stage. A remedy that sometimes works is to put a resistor of about 10,000 ohms between the grid of the first playback stage and whatever circuitry precedes this stage. As a further measure, try placing a small capacitance, about 5 or 10 pF, between this grid and ground.

Print-Through Problem

Q. When I record on Track 1 and then record on Track 2, I get print-through when I play back Track 1. The print-through is from Track 2. What does this mean?—H. E. Noles, Buffalo, New York

A. The "print-through" to which you refer may be adjacent channel cross-talk. And this may be due to improper vertical positioning of the tape head. In other words, the recordings for Tracks 1 and 2 may not be falling on the correct portions of the tape.

Trouble with 1/2-Mil Tape

Q. I have a tape recorder in the $350 price class. I am pleased with it but have found troubles using ½-mil tape. I am using two different name brands. After recording a great number of phono discs on these tapes, I find that on one of the brands of tape I get periods of silence. On the other brand I get a great deal of cross-talk, enough to ruin the sound of the recording. What is probably wrong? How can I correct it? How can I prevent these problems?—Hank Bernbaum, Hubbard Woods, Ill.

A. I cannot tell you how to "correct" your problems with ½-mil tape except to advise you to use 1- or 1½-mil tape. That people have problems—not always, but often—with ½-mil tape is perhaps not as well known as it might be. These problems include squeal, dropouts, high print-through (which may be the difficulty that you call cross-talk), stretch (and consequent distortion), etc. The NAB tape standard specifically states that ½-mil tape is "not recommended" except for the 3-inch reel containing 300 feet. I know of at least one manufacturer of high-quality machines who privately counsels that ½-mil tape should not be used with its machines.

Erasure and Noise

Q. I own a good, heavy-duty bulk eraser, and am thinking of disconnecting the erase head of my tape recorder and using the bulk eraser instead. This would involve nothing more than a switch and an equivalent impedance load to substitute for the disconnected erase head. Do you think that this would really make a noticeable difference in the background noise of my machine? — Thomas P. Witherspoon, Jacksonville, Fla.

A. Ordinarily the erase head adds no significant noise to a tape recording. It is the bias current in the record head that is responsible for appreciable noise. Hence there is no apparent advantage in disconnecting the erase head.

Taping Off TV

Q. I would like to tape off my TV. Is it better to connect the TV to my audio preamplifier or to the input of my tape recorder?—S. Valenza, Jackson Heights, N. Y.

A. In most TV sets, the audio signal present at the volume cotnrol is of high level, and therefore should be fed into a high level input, preferably that of your preamplifier. If you obtain the TV signal at the TV speaker leads, the signal level will depend on the setting of the TV volume control, which may at times be too low. Therefore it is preferable to take the signal off the hot and ground leads of the TV volume control, *provided the TV set has a power transformer to isolate it from the house line.* If not, then take the signal from across the TV speaker leads. Should the signal level then be too low in view of your customary setting of the TV volume control, you might feed the signal into the microphone input of your tape machine to obtain sufficient gain.

What Price Quality?

Q. I intend to buy a mono tape recorder with which I hope to record music. Being a student, I will have only about $100 to spend. Is it possible to get quality recorders in this price range?—Virgil Stubblefield, Rolla, Mo.

A. The meaning of "quality" necessarily varies from one person to another. Your ears may be more or less forgiving than another person's in the matters of distortion, frequency unbalance, noise, deviation from correct pitch, and wow and flutter. Still, there are some machines that record pretty decently at $100 or so. In such a machine, the sacrifices are not entirely in terms of performance; instead they may be in terms of operating conveniences and frills, which do not affect what you hear.

How Good Is Good Response?

*Q. I own a *** tape recorder with a claimed frequency response of 18 to 15,000 Hz, ± 3 dB at 7.5 ips, which isn't much. Lately I compared its sound quality with a new tape machine owned by a friend of mine, using a prerecorded tape, and I have noticed that the sound quality of my machine is definitely inferior. I would very much appreciate your advice.*—Duilio Armando Baroni, Belo Horizonte, Brazil.

A. Response flat within 3 dB to 15,000 Hz is not to be sneezed at. Such response is quite close to the NAB standard. However, your machine apparently is not living up to its specifications. A worn playback head is one of the likely reasons. Faulty playback equalization is another. Also consider the possibility that the fault may lie in your friend's machine—it may have over-bright response owing to incorrect playback equalization. Æ

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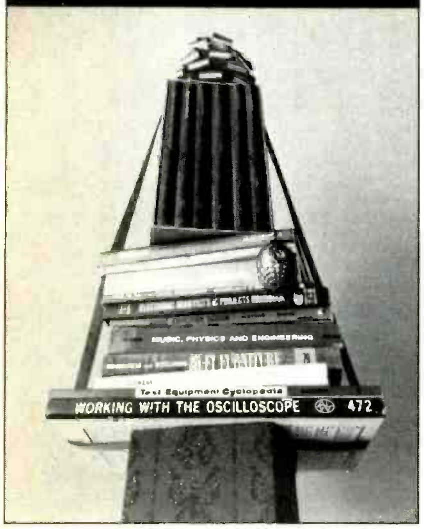

Many texts on the subject of high fidelity have been published over the years, some of which have become "classics." The parade of books continues, however. Here are some new ones that captured our interest this year. Some are for music lovers who wish to learn more about the subject. Others are more specialized, being directed toward experimenters, tape recordists, or servicemen. Whatever your leanings, however, there is probably a book here that you could use to expand your hi-fi reference library. (All are soft-cover editions unless otherwise noted.)

Encyclopedia of High Fidelity. Edited by John Borwick. Focal Press, Inc., New York, N. Y. Six Volumes, Hardbound. $49.50 boxed set; $9.50 each. This six-volume set covers the entire field of sound reproduction: "Amplifiers," "Acoustics," "Disc Recording and Reproduction," "Tape Recording and Reproduction," "Radio Reception," and "Loudspeakers." Experts in each field present basic theory and useful operating techniques for serious audio enthusiasts and technical students.

This is a grand undertaking. But, unfortunately, there are some major shortcomings in evidence. Published in Great Britain in 1963, the books do not include many technological advances made in the intervening years. And as Great Britain lagged far behind the United States in developing transistorized equipment, little space is devoted to solid-state components. Additionally, the few product examples given are of British manufacture, most of which are not sold here. Based on the foregoing, we cannot recommend the volumes, "Amplifiers" and "Radio Reception." Nonetheless, each of the other volumes has much to offer readers since, in British fashion, they cover the subjects in great depth.

Here are two paperbound texts on hi-fi, both revised editions, that might appeal to readers who missed the earlier editions: **Hi-Fi Loudspeakers and Enclosures,** 2nd Ed., by Abraham Cohen. Hayden Book Co., Inc., New York, N. Y. 438 pages. $5.95. This updated text, one of the classics in the field, covers an important subject in great depth. Divided into five sections, "The Loudspeaker," "The Enclosure," "The Room," "Stereo Practice," and "Acoutic Measurements" (plus enclosure construction details). Lucid explanations, backed up by many excellent drawings and photographs, should provide readers with a firm understanding of speaker principles, types, and applications. **Hi-Fi Stereo Handbook,** 3rd Ed., by William F. Boyce. Howard W. Sams & Co., Inc., Indianapolis, Ind. 288 pages, $4.50. This is a good, all-around book for beginners who wish to gain an understanding of hi-fi stereo equipment, as well as for more experienced persons who want to expand their knowledge of the subject. The author writes in a down-to-earth manner, covering hi-fi equipment — what each component does, how it works, etc. Updating includes the addition of several transistor circuits, though tubed equipment is still discussed in detail.

Music, Physics and Engineering 2nd Ed., by Harry F. Olson. Dover Publications, Inc., New York, N. Y. 460 pages, $2.75. Formerly titled "Musical Engineering," this revised and expanded book covers a wide range of subjects concerning sound reproduction, including sound waves, musical instruments, room acoustics, sound-reproducing systems, and electronic music (a 41-page chapter that has been added to the original book). There is a wealth of information here, most of which can be readily understood without special training in any of the arts or sciences. This marvelous book describes how instruments produce their sounds, for example; offers suggestions on microphone pickup arrangements; discusses how a digital computer "composes" music; and much more, as well as touching on "hi-fi" and "stereo."

Experimenters and hobbyists had a windfall in books this year. Here are some interesting ones: **How to Build Speaker Enclosures** by Alexis Badmaieff and Don Davis. Howard W. Sams & Co., Inc., Indiapapolis, Ind. 144 pages, $3.25. Wonderful how-to-do-it book, this one. It *could* have been a dull, wood-construction-plans text, but the authors packed it with solid, practical information on speakers and enclosures. Though directed to the home constructors, it's a worthwhile text for anyone who wishes to learn more about speaker systems. Includes nomographs, fine step-by-step construction photos, materials lists, and test techniques. **Electronic Hobbyist's IC Projects Handbook** by Robert M. Brown. Tab Books, Blue Ridge Summit, Pa. 159 pages, $6.95, hardbound; $3.95, soft cover. Covers 50 projects based on use of popular integrated circuits. Nearly half of the projects concern audio, including how to build a wireless mike, an FM radio front end, a portable public address system, and a 50-watt amplifier, among others. Projects do not include any photographs of completed equipment, nor does the author provide much guidance. Essentially, one is left with brief descriptions, a schematic, and a parts list. Therefore, it would be best if one had some experience in constructing electronic devices, to take full advantage of the book. A better book for tyros in construction and electronics is **Integrated Circuits, fundamentals and projects** by Rufus P. Turner. Allied Radio Corp., Chicago, Ill. 95 pages, 75¢. This get-acquainted-with-ICs manual lays a foundation for following construction projects by discussing what an IC is, how it works, and construction hints. Only six projects are detailed, including a high-gain preamplifier and a quarter-watt audio amplifier, but these are unusually complete, down to excellent photographs and drawings, as well as tips on testing out completed projects. **Designing and Building Hi-Fi Furniture** by Jeff Markell. Gernsback Library, Inc., New York, N. Y., 224 pages, $2.90, is a fourth printing of a unique book first published in 1959. Woodworking and finishing principles remain the same, of course, so that the book is not at all obsoleted by the passing years. **Audio Amplifier Design** by Farl J. Waters. Howard W. Sams & Co., Inc., Indianapolis, Ind., 160 pages, $3.25, illustrates how to apply theory in designing audio amplifiers. It's for the advanced hobbyist and for anyone who wishes an insight into what makes an amplifier tick. Both tubed and transistorized equipment are discussed. The style of presentation—each stage discussed in theory, followed by design methods using examples, concluding with a step-by-step solving of a design problem—is especially welcome.

Testing and measurement methods have been covered by a variety of texts, primarily directed toward hobbyists, servicemen, and professionals. Four books published by Howard W. Sams Co., Inc., Indianapolis, Ind., are most interesting: **101 Ways to Use Your Hi-Fi Test Equipment, 2nd Ed.,,** by Robert G. Middleton, 144 pages, $2.95, answers basic questions concerning tests of hi-fi equipment, including measuring bias current in a tape record head, checking a record player for rumble, and so on. Clear drawings, concise writing style. **Measuring Hi-Fi Amplifiers** by Mannie Horowitz. 159 pages, $3.25, combines theory with basic measurement methods. A little heavy on math, a basic technical background would be desirable to get full benefit from the book. **Acoustical Tests and Measurements** by Don Davis.192 pages, $4.95, is a long-awaited how-to-do-it text in the field of acoustics. The author covers acoustical instrumentation, testing methods and survey report writing. Chapters includes: How to Measure Reverberation Time, How to Measure Ambient Noise, and Testing the Sound System, among others. Well illustrated and nicely written. **Test Equipment Cyclopedia,** 104 King-Size pages, $1.95. A number of authors discuss how to use specialized test equipment for audio, radio, and TV. Includes chapters on Oscilloscopes for Multiplex, Decibels for Beginners, and Test Equipment Kits. Another "King-Size" manual, **Working with the Oscilloscope** by Albert C. W. Saunders, Tab Books,

(Continued on page 79)

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EDITOR'S REVIEW

Hi-Fi Standards

At a recent meeting of the Institute of Electrical and Electronic Engineers' Audio Group, Delaware Valley section, members expressed concern about the Federal Government's current moves on standards for consumer goods (e.g., method of measuring size of TV picture tubes, X-ray radiation of TV color tubes, and so on). How much longer will it be before the words "high fidelity" are defined, they wondered. Some felt that the "industry" should assume the responsibility of drawing the line between hi-fi and low-fi. A few, including ourselves, disagreed with the concept of such a sharp division, if only because it invites major compromises which would, in our opinion, give an aura of respectability to some "hi-fi" equipment that is too far removed from top-grade equipment.

As an example of what could happen, witness some guidelines that the German Institute for Industrial Standards has set as the minimum acceptable standard (DIN 45 500) with which the term hi-fi may be used: *Amplifiers.* Frequency response: 40 Hz to 16 kHz ±2 dB. Intermodulation distortion: under 3%. Power output (stereo): two 6-watt amplifiers. *Cartridges.* Frequency response: 63.5 Hz to 8 kHz ±2 dB; 8 kHz to 12.5 kHz ±5 dB. Channel separation at 1 kHz, 20 dB.

The above standards are certainly modest ones. In view of this, if any standards are to be applied, we would much rather see ratings related to levels of performance–four stars, three stars, and so on–that would truly separate the top-notch equipment from the medium-quality equipment from the low-to-medium quality equipment. But, then, this division is fairly well done in the component hi-fi field right now–by price.

Leinsdorf on Audio

Erich Leinsdorf, music director of the Boston Symphony Orchestra, guest speaker at the Audio Engineering Society's recent convention banquet, gave a spirited talk that was all the more admirable since it came at the end of interminable award-giving and award-acceptance speeches.

One of the themes voiced by Dr. Leinsdorf was the need for acoustic insulation to improve home sound reproduction and to shut out extraneous noise. Concerning the latter, he likened the acoustic environment in many homes to that of "Luna Park." Another theme was a pointed attack on the poor audio quality available from television receivers. He observed that WGBH FM in Boston alleviates the problem somewhat by broadcasting TV audio so that hi-fiers can listen to good quality sound on their hi-fi systems while viewing the video portion of a program on a TV set. Wouldn't it be nice if more FM stations did this?

* * *

Newly-elected officers of the Audio Engineering Society are: President: Benjamin B. Bauer, CBS Laboratories; Executive Vice President: Frederick V. Hunt, Harvard University; Eastern Vice President: George W. Bartlett, National Association of Broadcasters; Central Vice President: Harold O. Kaitchuck, Boulevard Recording Studios; Western Vice President: A. R. Soffel, LTV Research Center; Secretary: John D. Colvin, Commercial Radio Sound Corp.; Treasurer: John J. Bubbers, Pickering & Company, Inc.

Hi-Fi Shows

The 1968 San Francisco Component High Fidelity Show, sponsored by the Institute of High Fidelity, attracted a record 13,000 people, according to the IHF. Seminars held for the first time at the West Coast show were a huge success, too. Robert Moog, pioneer in electronic sound, drew over 500 people to his symposium on electronic music, for example. Stanton Magnetics was awarded the IHF's first annual "Best Dressed Hi-Fi Show Exhibit" award. (They were among the three exhibits we chose in the November issue of AUDIO as being the most attractive.) *A.P.S.*

Seasons Greetings from all of us to all of you.

The Commonality of Speaker Systems & Musical Instruments

ANTONY DOSCHEK

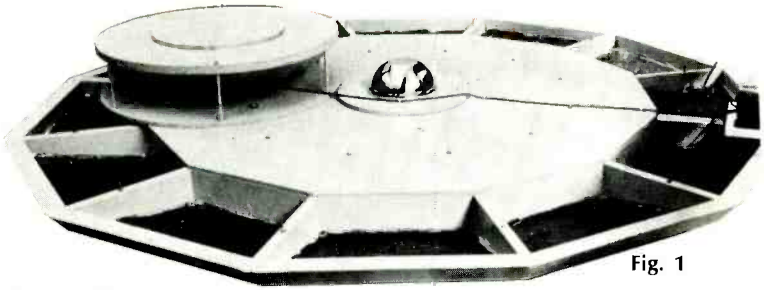


Fig. 1

Fig. 1—Radial transducer: an exponential horn system with flare expansion against a plane surface; 12-ft. diameter, 28-sq.-ft. mouth area works into semi-infinite space; an 18-in. bass driver, mounted horizontally, works into the throat annulus area via a concrete reflector cone mounted on the base plane, with a 300-Hz crossover; a 12-in. mid-range driver is similarly horn loaded, with crossover at 4000 Hz. The finished model used 6 horn tweeters (one shown) disposed at equal intervals around periphery. Double-thickness, 1/4-inch tempered Masonite flare segments are damped by 3/4-in. automotive muffle-coat. The bass driver back-load cavity is enclosed by a stainless-steel baker's mixing bowl.

PART I: Aerophones

THE ANALOGIES that can be drawn between man's artifacts and natural phenomena are legion. In a sense, man simply imitates nature. For example, every fundamental mechanism that creates the sounds of music already exists in nature: vibrating reeds, cavities, strings, membranes, slabs. Even electric guitar and organ pickups have a parallel in atmospheric "whistlers" and the "dawn chorus" so often disturbing to short-wave communications.

If it is possible to find analogies between instruments and nature, it may be interesting to examine acoustical comparisons between musical instruments and man's modern artifacts — loudspeaker systems. A discussion of vocal tone production would be a good starting point.

The principal acoustical factors relating to a singing voice are *range, color,* and *power*. Briefly, the human vocal system includes the lung and chest cavities, the wind-pipe, the larynx, and the throat, mouth, and nose cavities. Breath evhaled by the lungs, plus a neuro-muscular volition to create sound, causes the vocal cords to vibrate at a rate determined by the degree of tension under which they are maintained. This, in turn, determines the fundamental pitch of the sound that is produced. On an average basis, the fundamental speaking-voice pitch for women is about 250 Hz. (near middle-C on the piano keyboard) and about one octave lower for male voices. The usable range of a trained singing voice is normally two to two-and-one-half octaves, while a few exceptional voices can manage a range of three-and-one-half octaves, and more.

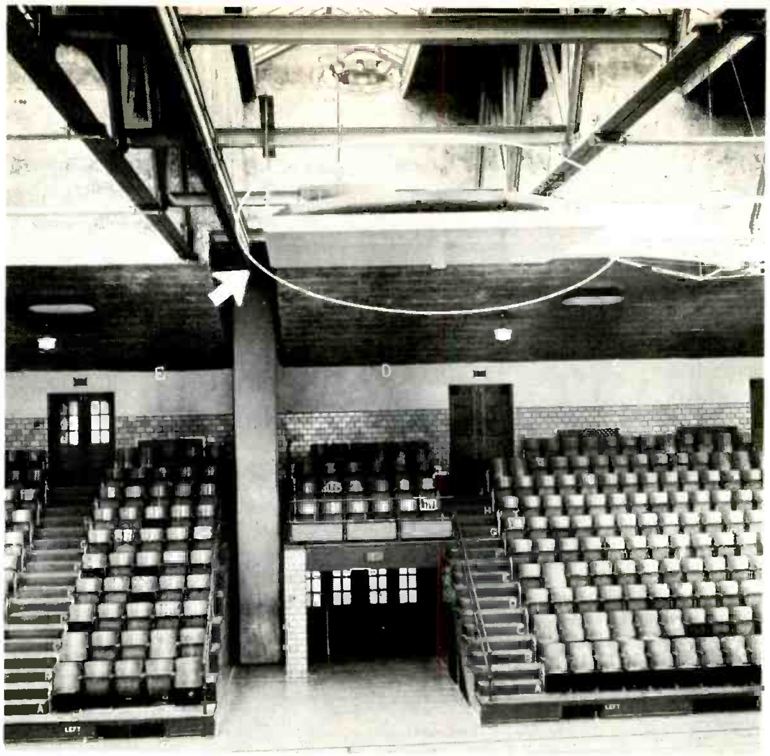
The color of a singing voice is determined by a number of factors. Primarily, the vocal cords do not vibrate in a smoothly symmetrical manner to produce a pure, sine-like tone similar to that of a tuning fork. Instead, their tonal output approximates what is called a sawtooth wave; one in which each cycle shows a rapid rise from minimum to maximum and a slower decay back to minimum. Such a wave consists of the fundamental frequency and both even- and odd-order harmonic overtones. If we simply multiply the fundamental frequency of a tone by 1, 2, 3, 4, 5 . . . up to the highest audible frequency in that tone, we get the entire even- and odd-order series. In a sawtooth wave all the harmonic overtones diminish regularly in amplitude so that the very high orders are both too weak to be audible, as well as being beyond the normal range of human pitch perception. But a well-trained singer can modify the motion of the vocal cords and alter the harmonic structure of their output waveform to produce a round, mellow quality or a piercing tone, as well as many shades between. On the other hand, the harmonic-rich sound output of the vocal cords takes place well within the respiratory tract, which bears an acoustical resemblance to a closed-end organ pipe, and such a structure resonates only in the odd-order harmonic series. Therefore, we might expect the singing voice to contain both even and odd harmonics, but be especially resonant in the odd-order series. The distaff contingent upsets this concept. It happens that women's voices show the most prominent resonance one octave above the fundamental (even-order 2nd harmonic), while men's voices are inclined to support expectancy with a principal resonance one octave-and-a-fifth above the fundamental frequency (odd-order 3rd harmonic).

A second factor contributing to the color of the voice is the length and shape of the several distinctive cavities comprising the vocal tracts, and how they are altered in size and shape by the ability of the singer. The position of the lips, the placement of the tongue, and the degree of throat construction have the most profound effect on tonal color. This can be demonstrated clearly by noting the position of the lips and tongue while vocalizing the sounds "ee" and "oo." In addition, there are about forty "voiced" and "voiceless" speech sounds that contribute individual shades of color to the singing tone.

The power of a singing tone is largely a matter of physical endowment modeled by intelligent, inten-

Fig. 2—The base plane of a radial transducer is lifted into position. 15-ft. square by 9-in. thick, framed and cross-braced by 8-in. x 2-in. beams, and clad with 3/4-in. plywood. Internally, it is filled with vermiculite and undercoated. Projections you see are 1/2-in. steel-threaded studs and stand-off spacer bushings to support the horn flare.

Fig. 3—The **radial transducer** is suspension-mounted in combination gym/auditorium of a large high-school; total weight is over 2600 pounds. Recordings of "Echoes of the Storm," Stravinsky's "Sacre . . . ," and the "Sibelius 2nd" were used during check-out proceedings. Students organized hi-fi dances on Friday evenings, paying for the entire installation in less than one school-year. ▶



sive training; although the color of the voice and the dynamic variation imparted by the singer also contribute to the listener's subjective sensation of power. A still further contributing factor to the power of a voice is the concept of *purity*. Purity is a characteristic of tone defined by an aesthetic or artistically traditional standard which cannot be described in engineering terms. But the physical basis for voice power is simply the amount of air (volume current) that the singer is able to force past the vocal cords, and the strength of the resonances in the vocal tract.

The color and transient quality of speech, and the tonal simplicity of lightly accompanied song, make these sources particularly informative in the evaluation of speaker system performance. (More about this later.)

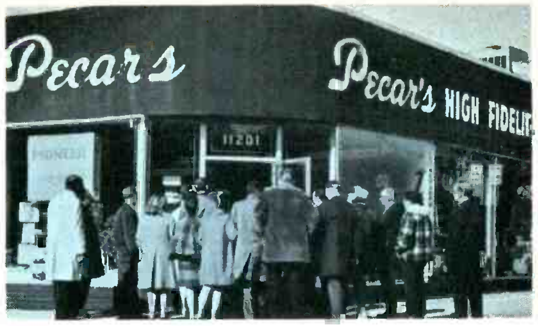
The entire family of wind instruments, the aerophones, shows a close acoustical relationship to the human voice because air forced past the lip-controlled reeds of the woodwinds and the fluttering lips of the brass players creates complex tones which are resonated within the tubulations of these instruments. Changing the length of the tubulations by successive opening or closing a series of holes along the tube or by valving the air column into auxiliary lengths changes the pitch of the sounds. As a matter of fact, the family of woodwinds consisting of the oboe, English-horn, bassoon, and sarrusophone all have a double reed that behaves much like two vocal cords, and a conical bore with a rate of flare somewhat similar to that of the human respiratory tract. The associative appeal of the double-reed tone is striking: note particularly the English-horn solos in Dvorak's *New World* or Sibelius' *Swan*. The single-reed instruments of the clarinet family show acoustical spectra about as rich in harmonics as the double reeds. These instruments have conical bores, rather than cylindrical, and produce a different amplitude distribution of their odd-order harmonics. In addition, they sound the fundamental in a lower register than the conical bores do. The saxophone family of single-reed/conical-bore instruments generate a still different tone color than their more classical brethren because the reeds and mouthpieces are larger and the flare constants expand more rapidly.

There is a unique analogy here in the air-horn speaker: an interesting hybrid that incorporates an electromechanical throat valve modulated by a vocal or musical signal. Air under high pressure is forced through the valve which feeds an exponential or hyperbolic horn to produce enormous sound pressure levels for disaster alarms and long-distance airborne communication.

The brasses, called lip-reed instruments, have bores that are combinations of both conical and cylindrical flares. Since the lowest pitch that a pipe can produce depends upon its length, and the harmonics that can be sounded are relatively few in number for any given length, brass instruments must be provided with rotary or piston valves and tube extensions (called crooks). Furthermore, all this plumbing must be coiled ingeniously so that it can be held in some relative degree of comfort because, on an average, bass tubas use about eighteen feet of tube, French-horns twelve feet, trombones nine feet, and the trumpets or cornets about six feet. The tone color of the brasses depends a great deal upon the shape and depth of the mouthpiece cup as well as the size and sharpness of its outlet orifice. These factors all modify the back-pressure upon the player's lips and thereby affect the harmonic content of the tone.

Other obvious analogies to the acoustical behaviour of the voice and wind instruments are to be

the crowd pleaser...



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found in the reed pipes of the organ, the harmonica, accordions, etc. But less obvious is the fact that flutes and piccolos, recorders, various kinds of whistles operate in the manner of reed instruments. Air passing into the fipple-hole or embouchure behaves somewhat like a reed in itself; academically, these are known as air-reed instruments. As air enters the bore of the instrument it builds up a pressure to the point where the air stream is deflected across the embouchure lip and back out of the bore, whereupon the internal pressure dies down and again allows the air stream to enter the instrument. The marvel of this is that it happens as many times as there are cycles in the fundamental pitch being sounded by the player: in each second of time, and for every second that a flutist holds on to his high C, for example, his breath flips in and out of the flute 2093 times!—and almost twice as fast for the piccolo player's high B-flat. Talk about the fastest gun in the West!

Aside from the air-horn there are no speaker systems directly comparable to the voice or aerophones in that cone vibrations modulate a *flowing* stream of air. However, in the "olden days" of hi-fi, when bass response was very hard to come by, there was a popular speaker system known as a *bass-coupler*. The bass-coupler consisted of a long, rectangular enclosure driven at one end by a cone loudspeaker venting through a slot aperture. The arrangement resonated as a closed-end pipe (similar to the organ bourdon) and showed strong amplification of the loudspeaker program in regions of its fundamental frequency and the series of odd harmonics. If the coupler was eight feet long, it had a whopping big drone at 35 hertz and a nice assortment of honks at 105, 175, 245 . . . hertz. Though the very low bass was amplified, the listener soon discovered that most of the sound in that region was turntable rumble and groove grind. Several variations of this scheme were devised with the aim of smoothing out the ragged response, and one that went by the name *corner-coupler provided* a smoother characteristic because the long, closed-end triangular pipe was designed to be a bass-reflex enclosure with a tuned and critically damped port. This version showed pleasant, wide-response reproduction at low-listening levels for background music in restaurants and other public places.

Of course, the family of horn speakers is closely related to the woodwinds and brasses in structure, but here the acoustical properties of the horn are designed to function in a different manner. Instead of utilizing the resonances of the horn as musical instruments do, loudspeakers use the horn to translate acoustical energy smoothly from a small throat area at the driver cone or diaphragm to the relatively large mouth area of the horn. The loudspeaker horn, in principle, is an acoustic transformer which converts the high-velocity/low-volume of air at its throat to a low-velocity/high-volume at the mouth, thereby feeding the sound wave into the listening room more efficiently. Furthermore, if the mouth area is made large enough relative to the frequency range that it is intended for, the horn shows no prominent resonances. The flare constant of a horn—both the manner and rate of its expansion—determines its efficiency as well as any distortions that it is capable of producing (which are due to the non-linear compressibility of the air itself). Many types of horns have been studied, the most common ones being the parabolic, the conical, the exponential, and the hyperbolic. The terms refer to the mathematical formulas governing the rate of flare. Taking all factors into consideration, the exponential horn embodies the best practical combination and is, therefore, the one most commonly used for music reproduction.

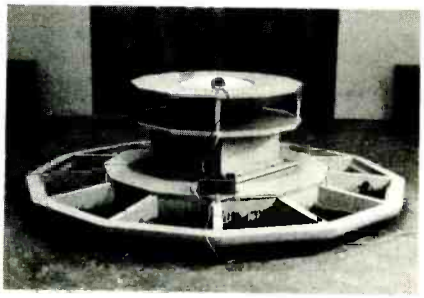

Fig. 4—Here's a radial transducer, junior version. Its dimensions are 8-ft. diameter, 14 sq. ft. mouth area. With construction similar to the 12-ft. diameter version, it contains a 15-in. bass driver, 8-in. midrange, and a horn tweeter.

The exponential horn shows a number of scientifically provable advantages and a few practical disadvantages. Because of its high electrical-to-acoustical energy conversion efficiency, the driver cone or diaphragm is not required to make large back and forth excursions in order to generate high sound pressure levels in the listening room. This minimizes the frequency modulation (Doppler) distortions that occur when the low-frequency movement of the cone is a significant portion of some high-frequency wavelength which the cone may be reproducing at the same time. Doppler modulation causes an unclear midrange sound and, in severe cases, a clearly audible warbling in the high-frequency part of the program. The good efficiency of horns also requires much less amplifier power with the resultant improvement in the overall signal-to-noise ratio. Listening tests have demonstrated that a small amount of purposely introduced harmonic distortion in quiet solo or small ensemble recordings was less injurious to the illusion of realism than a similar amount of hum or random noise. A further desirable effect of horns on music reproduction lies in the sense of greater sound dispersion attributed to the large area of horn mouth. The sound of a good horn reproducer system is characterized by a certain effortless quality connoting virtually limitless power capability and a pervasive distribution of sound.

The disadvantages of wide-response horn speaker systems lie in their cost and size. To be practical for home or building installation, large horns use a "folded" air column design. This is expensive to carry out in manufacture and must be combined with additional mid- and high-frequency horns and costly frequency - distribution networks. Furthermore, their best placement is found to be in a room corner (adjacent corners for stereo), and these are not always available. Yet no serious music enthusiast argues with the size and ungainliness of a grand piano!

(Continued Next Month)

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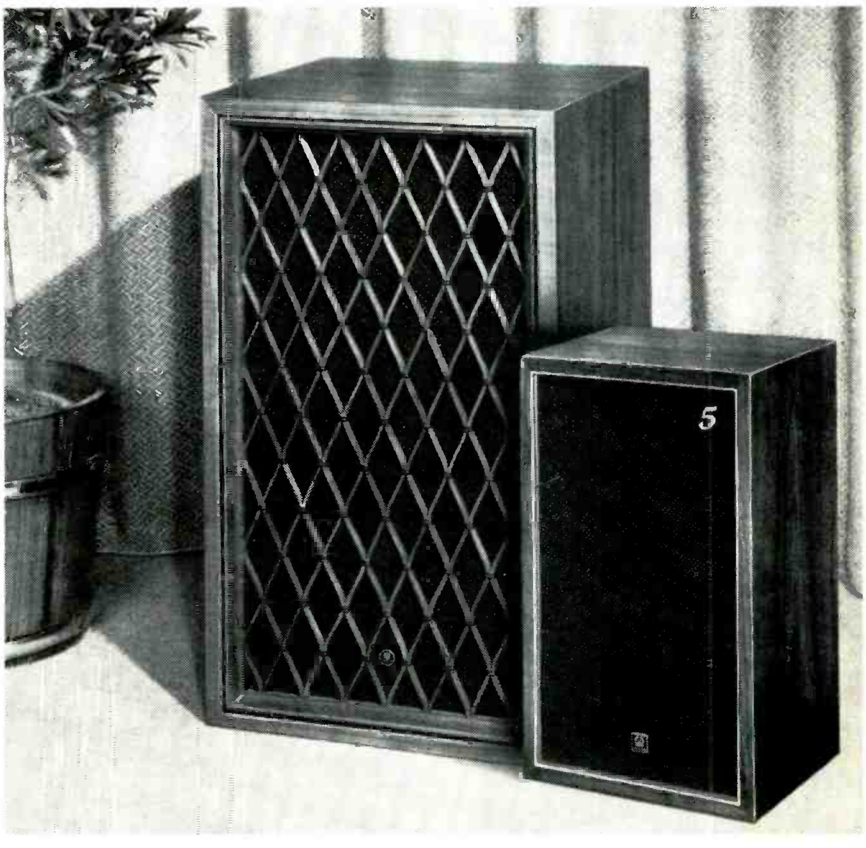

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The 20 most-often-asked Questions about Hi-Fi [with Answers]

LEONARD FELDMAN

At the recent New York High Fidelity Show sponsored by the Institute of High Fidelity I had the pleasure (for the fifth year in a row) of speaking to audiences on four successive days. My topic was "Introduction to High Fidelity Components." As is my usual practice, I welcomed questions from the audience after my remarks, answering them as best I could. I found that the questions were not only of great interest to the newcomer to hi-fi components, but seemed to crop up repetitively each day with each new audience. Indeed, some of these questions keep repeating from year to year, from one symposium to the next.

Here, then, are the twenty most-often-asked questions which were put to me, along with my answers as nearly as I can recall them. The questions, of course, covered every phase of hi-fi componentry and were "thrown" at random. I have re-arranged them into six categories of subject material in the interest of organizing the material.

Tuner Questions

Q. *You say that a separate outdoor FM antenna should be used for good FM and stereo FM reception. I'm not permitted to install a second outdoor antenna. Is there any way I can connect from my TV antenna to my FM set and how bad is this arrangement?*

A. While TV antennas are cut in length for best TV reception, they are not at all far from the correct length for use with FM tuners or receivers. Generally, all you need do is connect a piece of "twin-lead" 300-ohm cable from the two TV antenna screw terminals to the two FM antenna screw terminals on the back of your tuner or receiver.

There are a few disadvantages to this, other than the fact that the antenna is not exactly the ideal length and design for best FM reception. For example, if you play the TV set and the FM set simultaneously, certain combinations of FM frequencies and TV channel frequencies can cause interference from one piece of equipment to the other. This is usually not a great problem, since in most instances only one appliance will be used at any given time. No *physical* damage will be caused to either piece of equipment with the interconnection described. Because of the slight impedance "mismatch" with two such appliances connected, there will be some signal-strength reduction for both sets of equipment. In the case of the TV set, however, TV signals are usually more than adequate in most metropolitan locations and, as far as FM reception is concerned, it will definitely be better than if an indoor "rabbit-ear" or a piece of wire behind the set were the only antenna used. If you are plagued with "multipath" distortion (a form of distortion in stereo FM analogous to "ghosts" or reflections on TV) it may be necessary to arrive at some compromise orientation of the common antenna so that it is satisfactory for both TV and FM service. Alternatively, you might use a two-set splitter although this will reduce signal strength to *both* TV and FM sets. Of course, for truly optimum reception you should use an antenna expressly made for FM reception.

Q. *I own a *** preamplifier-amplifier combination purchased several years ago. It is a tube-type set. Now I want to purchase an FM/AM tuner. Must I look for a vacuum-tube type or can I buy a modern, solid-state tuner?*

A. The nice thing about stereo high fidelity components is the nearly total absence of any "built-in obsolescence." While solid-state circuitry (that is, equipment containing transistors instead of vacuum tubes) is radically different from its tube predecessors, the inputs and outputs are perfectly suited in signal level and impedance for use with any vacuum tube associated equipment. Incidentally, the converse is also true. For example, a tube tuner will work well with a solid-state amplifier. Also, either tube or transistor amplifiers will work well with nearly every high fidelity loudspeaker system —be it of recent or of ancient vintage.

Q. *What is meant by the specification, "Capture Ratio"?*

A. This specification is used to describe, numerically, the ability of an FM tuner to "capture" or favor the stronger of two FM stations broadcasting at the same frequency. You might suppose that such a rating is rather academic in view of the fact that stations are not assigned the same frequency in a given locale, but consider the following possibility: Suppose you like to listen to a relatively weak station broadcasting right in your home town or suburb, and assume that this station is received with a signal strength of 50 microvolts at your antenna terminals. Now, let's say that about two hundred miles away there's a real "powerhouse" of an FM station which, with present-day, ultra-sensitive tuners and good antennas reaches your set at a signal intensity of, say, 15 microvolts. Will you hear the desired station, the undesired station or, what's worse, both stations at once?

Capture Ratio, expressed in dB (decibels), tells you how much stronger the desired signal must be compared to the undesired signal in order to "block out" the undesired signal to a point where it is just barely audible. The *lower* the dB figure for capture ratio, the less the differential between undesired and desired signal needs to be. Obviously, this specification can never reach "0 dB," for this would mean that the tuner has a mind of its own and can select the "desired" signal, even though it is exactly equal in strength to the undesired signal.

Q. *What is meant by "crystal i.f. filters" and "ceramic i.f. filters," and in what way, if any, are they superior to conventional i.f. transformers?*

A. Up until recently, the i.f. system of an FM set consisted of several stages of amplification interspersed with tuned circuits in the form of transformers (with resonating capacitors across both primary and secondary). The purpose of the transformers was to establish the "bandwidth" or pass-band of the i.f. system. In standard FM, this bandwidth needs to be at least 150 kHz, since audio modulation of the carrier causes a change of frequency of a maximum of ± 75 kHz about the center frequency. To get such a bandwidth with all portions of it having the same

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How to flip over the sound without flipping over the reel.

For years you've flipped over a little thing called a reel. You flipped because the tape would run out at inconvenient times like 30 seconds into the Minute Waltz, or three and a half movements of your favorite symphony.

Something had to be done about it.

Panasonic, the world's leading manufacturer of tape recorders, did something. That something is the Panasonic Symposium. It's a Solid-State 4-track stereo deck that's unflippable.

Unflippable because of continuous Automatic Reverse. You'll never flip over another reel again. And the turn around is so quick you'll hardly miss a beat.

And the beat is steady. That's because there's Dual Capstan drive on all three speeds. That way the Tijuana Brass won't sound like 76 Trombones and vice versa.

Of course, if you don't want it to run forever, use the automatic shutoff.

Will the Symposium match your rig at home? Yes. It has 30-20,000 CPS response, and the signal-to-noise ratio is more than 52 db's plus a recording system that has an AC bias of 90 kc. For sound-*on*-sound or sound-*with*-sound that's plenty of fidelity.

You'll be crazy about our control panel features, too. Like headphone output and Pause Control for easy editing. Then there are two big VU meters, each sensitive enough to catch the difference between a wheeze and a whisper. Plus a 4-place digital tape counter for some of that long-distance taping you might get involved in.

Top it all off with a smoked-glass dust cover that doesn't cost extra—and everything's beautiful.

So why not go down to any dealer we permit to carry the Panasonic line. We think that once you hook up our Model RS-796, you can stop flipping over its reel and really start flipping over what you hear.

Tape Deck, Model RS-796

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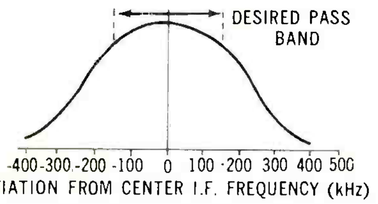
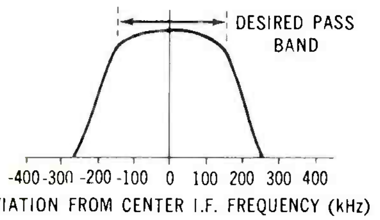
amplification was well nigh impossible, using transformers.

The bandwidth response of a typical i.f. system is as shown in Fig. 1, when transformers are used. While the relative amplitude over a bandwidth of 150 kHz is almost constant, to get this result, the total bandwidth has to be much greater, say, 400 or 500 kHz. This means relatively poor *selectivity* — or lack of ability to pick out one station from its frequency-adjacent neighbor. With crystal or ceramic filters in place of conventional inter-stage transformers, it becomes possible to achieve bandwidth characteristics that look somewhat like that shown in Fig. 2—or closer to the ideal of a straight or flat-top portion for 150 to 200 kHz and then an almost immediate falling-off of amplitude beyond these end points. Such arrangements promise improved selectivity in today's ever more crowded FM band.

Fig. 1 (top)—Typical response of conventional i.f. system.

Fig. 2—More nearly ideal i.f. response attainable with crystal ceramic filters.





Amplifier Questions

Q. *Why are there two types of power ratings for amplifiers and what is the difference between "r.m.s. or continuous power" and IHF or "music power"?*

A. R.m.s. or continuous power rating refers to an amplifier's maximum power-handling capacity when fed with a constant tone, such as a pure sine-wave. The power output is measured under conditions of ever-increasing input signal, until "rated distortion" is reached. What the rated distortion shall be is left up to the manufacturer, so it is important to consider this part of the "spec" as well as the absolute number of watts listed. Thus, "20 watts of continuous power at 1% distortion" is not as good as "20 watts of continuous power at 0.5% distortion."

Most listeners, of course, do not listen to shrill sine waves, preferring music! Under conditions of musical input signal, most amplifiers will be able to produce somewhat more power (because they are not being called to deliver maxima for anything but a short duration) than under "continuous signal" conditions. This fact gives rise to the music-power system of power ratings for amplifiers. While there is no direct correlation between an amplifier's continuous power rating and its music power rating, the latter may be anywhere from 15 to 50% greater than the former. The difference will depend upon the stability and ruggedness of the internal power supply of the given amplifier. The more rugged this power supply, the less the difference between the two ratings. The important thing to remember is to compare similar forms of specifications when comparing amplifiers, in fairness to the products under consideration.

Q. *If, as you say, the ideal for an amplifier is absolutely uniform or flat response from 20 Hz to 20,000 Hz, why do most amplifiers (or at least preamp-amplifiers) come equipped with wide-range bass and treble controls which enable us to upset this flat response?*

A. The ideal situation, acoustically, is flat response over the entire range of audible frequencies—but notice I said *acoustically*. Often, a heavily draped and carpeted room may be overly absorptive of high or treble tones. Also, your loudspeakers may be somewhat deficient in bass-reproducing capability, either because of their own design limitations or because of poor placement in the listening area. In such cases, *moderate* use of treble and bass tone controls helps to compensate for these deficiencies in other parts of the system. Further, the user may wish to change the tonal balance established by the recording engineer.

Q. *If solid-state designs have now made possible high-powered, all-in-one-chassis integrated receivers, why do some manufacturers still offer separate preamp-amplifiers and even separate preamplifiers and separate amplifiers?*

A. The high fidelity component industry started with "separates" nearly two decades ago. It was felt that the use of a separate tuner, a separate preamplifier, and even a separate controlless power amplifier (capable of being "hidden" out of view) offered the greatest measure of flexibility and quality. Many purchasers felt that they wanted to apply the maximum budgeted amount of money to the best amplifier or preamp-amplifier possible. This, with a record playing facility (turntable or record changer) and a pair of speaker systems constituted the start of the system. The tuner could be purchased at a later date, as finances permitted. Such thinking still abounds today and probably always will, so that manufacturers continue to offer either separates or all-in-one receiver chassis.

It should be noted, too, that the purchaser of separates generally seeks the ultimate control flexibility. For this reason, more secondary control features are usually incorporated in separate amp-preamps than in complete receivers. Finally, the really powerful amplifiers (over 100 watts per channel) needed for complex multiple speaker set-ups and semi-professional work do not appear in abundance among complete receiver design, but are plentiful among the "separates." Whether to assemble a system using a receiver as the electronic "center" or whether to do it by means of a separate tuner and a separate amplifier, therefore, depends upon requirements of power, extreme flexibility and, ultimately, personal preference.

Q. *How many sets of speakers can be connected to a stereo high-fidelity component amplifier?*

A. This depends upon many considerations. First, will you want all sets of speakers playing simultaneously? Then, how efficient are the proposed loudspeaker systems—or, to put it another way, how much power does the manufacturer recommend to be fed to each speaker for reasonable sound level? Finally, how loud do you like your music to be? The first thing you should realize is that with two sets of speakers going (assuming they are of equal nominal impedance, such as 8 ohms or 16 ohms), the amplifier power will be divided in half power to each speaker; with three sets, it will divide into thirds, and so on.

In the case of solid-state amplifiers used with multiple sets of speakers, there is another thing you need to watch out for. Most solid-state amplifiers should not be used with "load impedances" of less than 4 ohms, if damage to output circuits (or fuse blowing or circuit breaker popping) is to be avoided. Since speakers connected in parallel act just like resistors in parallel (as far as net impedance is concerned), this works both in favor and against the consumer intent upon having multiple speaker set-ups. True, a pair of 8-ohm speakers in parallel across an amplifier output channel will "look like" a 4-ohm load, the lower limit. It's also true, though, that most solid-state amplifiers deliver their greatest amount of undistorted power

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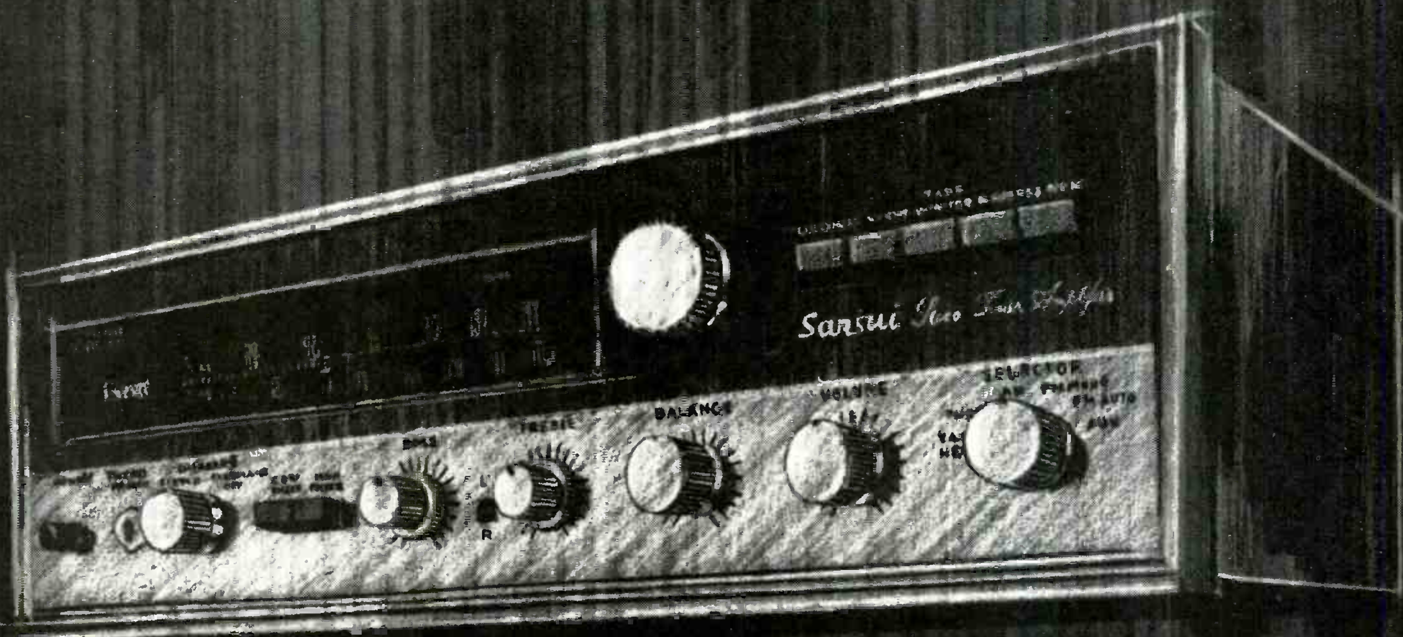

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Sansui Electric Company, Ltd. Tokyo, Japan • Electronic Distributors (Canada) British Columbia when feeding a 4-ohm load. Going further, however, if you were to connect *three* sets of loudspeakers across such an output channel, you'd be looking for trouble: the net impedance would be 8/3, or about 2.6 ohms, dangerously low for most solid-state amplifiers. In such cases, you'd be better off looking for 16-ohm speakers, in which case you could connect as many as *four* sets before reaching the four-ohm limit..

Receiver Questions

Q. *Why do some integrated receivers come with AM radio built in, while others only have FM and FM stereo?*

A. High fidelity stereo receivers, as the name implies, are intended for reproducing minimum-distortion, full-frequency-response music in your home. AM radio, as generally broadcast in this country, is far from a "high fidelity" medium, by our definition. Most AM stations seldom transmit frequencies above 5000 Hz (though they have the capability of doing so), whereas FM stations are required to transmit all frequencies from 50 Hz to 15,000 Hz—almost the entire range of audible sound. In addition, AM is subject to man-made and natural interference referred to as "static." Noise interference is more noticeable at high frequencies than at low frequencies and, because the amplifier portion of the component receiver has good response out to 20,000 Hz and better, such interference encountered on AM may actually be more annoying to the listener on a good-quality receiver than on an inexpensive table model radio having limited frequency response.

Still, as with all things relating to consumer opinion, some manufacturers offer the option of AM at a relatively minor increase in price. So, if there are some stations that you favor which are available only as AM outlets, the extra cost is not great. Nearly two years ago, stations which formerly broadcast the very same material on their AM and FM transmitters were forced, by the FCC, to offer separate programming at least 50% of the time they are on the air. This has resulted in an increase in popularity of AM, for we can no longer say, as we used to, "Why get AM when you can get most of the same programs on the FM counterpart?" The choice is really up to you.

Q. *Can all-in-one receiver performance specifications really be as good as the combined specifications of separate tuners and separate amplifiers?*

A. Indeed they can and sometimes are. Time was, when it was nearly impossible to build high-powered amplifiers on the same chassis with sensitive tuners. In the vacuum-tube era, the heat produced by large power output tubes precluded construction of complete receivers of power ratings much greater than 20 watts or so per channel. Now, with cooler-running transistors, there is no limit to performance or power possible in an integrated receiver. Bear in mind, too, that the use of a common chassis, a common power supply and other common parts makes for some economy in price with no real sacrifice of performance.

While there is no reason that a complete receiver cannot have exactly the same specs as "separates," many manufacturers, in the interest of further economy and mass acceptance, sometimes leave out some of the less important control functions associated with "separates." It is this design philosophy, right or wrong, which induces many purists still to prefer "separates." While we're on the subject, the same question might be asked of the new "compacts" (in which not only an integrated receiver is offered, but it is combined in a small cabinet with a record changer or turntable and arm and a pair of loudspeaker systems of the manufacturer's own choosing). Here, too, there is no reason why good specs cannot be built in. There are, however, a couple of extra pitfalls. Often, you may not agree with a particular manufacturer's choice of speakers or changer and cartridge offered in the compact, in which case you should turn to the component approach. In addition, acceptance of the "compact" approach has been so great that many "marginal quality" manufacturers have entered the field; the range of quality available in this area extends all the way from "lo-fi" to moderately good "hi-fi."

Speaker Questions

Q. *Are loudspeaker systems better off on the floor or above the floor by some distance?*

A. Generally, that depends upon the loudspeaker system. The smaller, bookshelf types will suffer little or no degradation in bass performance when elevated above the floor, as on a bookshelf. In fact, with the speaker at "ear-level," the otherwise directional *high* frequencies may reach you more effectively. Larger speaker enclosures are generally designed for positioning on the floor and, more importantly, will exhibit better bass performance if placed in a corner of the room rather than centered on a wall.

Q. *Are there any "speaker power ratings" similar to amplifier power ratings?*

A. Unfortunately, standards with respect to loudspeakers are not nearly as inflexible as they are with amplifiers or tuners. Some speakers list no information as to power ratings. Others give recommended minimum power required, while still others state maximum power ratings—or the amount of power which should not be *exceeded,* if the life of the speaker is to be preserved. This much can be said, in a general way. If a speaker manufacturer states that you should have available *at least* 20 watts of power to "drive" his product properly, you may safely assume that power even four times as great as this minimum will not damage the voice coil or cone suspension. If a manufacturer states that his speaker can handle a maximum of 50 watts of continuous power, then you can assume that it will handle double that amount for short periods of time, as might be encountered in musical crescendos, etc. Finally, if a speaker manufacturer offers absolutely *no advice* on the subject, don't take chances. Write to the maker directly or consult a knowledgeable hi-fi dealer.

Q. *You stated earlier that paralleling speakers may cause too low an impedance for the amplifier output circuits. How about wiring speakers in series? Doesn't that increase the impedance?*

A. Yes, wiring speakers in series does increase the net impedance. That is, two four-ohm speakers in series add up to 8 ohms and so forth. However, most speaker manufacturers frown upon this arrangement for two reasons. If you connect two dissimilar speakers in series, they no longer perform independently. That is, the action of one can actually influence the performance of the other. This is not true in parallel connections. Secondly, loudspeakers connected directly to both terminals of an amplifier "look back" into a very low amplifier impedance. The ratio of the speaker impedance to the amplifier's internal impedance is known as the "damping factor" and a high ratio helps to provide what has been variously called "stiff bass," tight bass or good transient response. With two speakers connected in *series,* each speaker "looks back" through the high impedance of its fellow speaker, providing a maximum damping factor of only 1, which is not considered adequate. The only time we suggest wiring speakers in series is if there is no other choice available and if the two speakers are identical models—and even then, with some reservations.

STANTON 681EE CALIBRATION STANDARD

"The tracking was excellent and distinctly better in this respect than any other cartridge we have tested....The frequency response of the Stanton 681EE was the flattest of the cartridges tested, within ±1 dB over most of the audio range."

From the laboratory tests of eleven cartridges, conducted by Julian D. Hirsch and Gladden B. Houck, as reported in HiFi/Stereo Review, July, 1968.

To anyone not familiar with the Stanton 681, this might seem to be an extraordinary statement. But to anyone else, such as professional engineers, these results simply confirm what they already know.

Your own 681 will perform exactly the same as the one tested by Hirsch-Houck. That is a guarantee. Every 681 is tested and measured against the laboratory standard for frequency response, channel separation, output, etc. The results are written by hand on the specifications enclosed with every 681.

You don't have to be a professional to hear the difference a Stanton 681 will make in your system, especially with the "Longhair" brush that provides the clean grooves so essential for flawless tracking and clear reproduction.

The 681EE, with elliptical stylus, is $60.00. The 681T, at $75.00, includes both an elliptical stylus (for your records) and an interchangeable conical stylus (for anyone else's records). For free literature, write to Stanton Magnetics, Inc., Plainview, L.I., N.Y.

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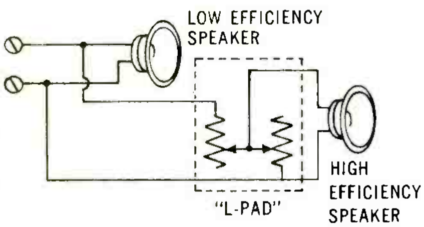
20 Questions

Q. *Is it ever necessary to use "volume controls" with speakers?*

A. Speaker level controls, usually referred to as "pads," serve a useful purpose in multiple speaker installations. Suppose, for example, that you have a low-efficiency bookshelf speaker system in your main listening area and have added a high efficiency, horn-type outdoor speaker for the patio. If you were to adjust the volume of sound for your main listening area, you might well find that it would be much too loud for the secondary listening area, because of the great difference in efficiencies between the two types of speakers. In such a case, an "L" pad, shown schematically in Fig. 3, should be wired into the circuit of the "patio" speaker. As you can see from the diagram, the L pad has the advantage of presenting a constant load impedance to the amplifier, while at the same time varying the sound level of the loudspeaker. Such pads come in different impedance ratings, such as 4, 8 or 16 ohms, to match the impedance of your speaker. Make sure, however, that the control you choose has sufficient power rating, for some of the power "taken away" from the loudspeaker to reduce its sound level must be dissipated in the resistive windings of the L-pad.

Fig. 3—An "L Pad" used as a speaker "volume control" when speakers of different efficiencies are used together. It may also be used as a remote-speaker volume control, of course.



Tape Equipment Questions

Q. *What is the difference between a so-called "Tape Deck" and a Tape Recorder?*

A. A tape recorder is often assumed to consist of a tape transport mechanism, as well as a complete recording amplifier (for making tape recordings) and a playback amplifier and speakers, all contained in the one package. Usually, because of size limitations, the amplifier and speaker (or speakers, if it is a stereo model) will be minimal-quality items, hardly consistent with hi-fi requirements. If the high fidelity component system owner is not interested in taking a recorder to another home, it makes sense to purchase a "tape deck," which contains all the elements mentioned above except the power amplifier(s) and speaker(s) for playback. With a tape deck, all you need to do is connect a pair of output cables (in the case of stereo) from the deck to an unused auxiliary input pair of jacks on your hi-fi amplifier or receiver. This enables you to listen to all your tapes through your better amplifier and speakers. To make recordings of anything but live sound (for which microphones are usually supplied), you need only connect another pair of cables from the "recorder output" jacks of your amplifier to the "high level inputs" on your tape deck. By so doing, you can directly record anything being played on your system, such as AM, FM, stereo FM, and Phono; no microphones are required for this procedure.

Q. *Are "cassette" type tape recorders capable of true high fidelity performance?*

A. In our opinion, not *yet*. Slow speed ($1\frac{7}{8}$ in. per second) combined with narrower tape width reduce the frequency response capability to approximately 8 to 10 kHz—good, but not quite good enough. The best signal-to-noise ratio we've measured so far is about 42 dB—again, good but not quite good enough. Cassettes do, however, offer many advantages. Ease of handling compared to "open" reel-to-reel systems, capability of recording as well as playback when compared to the $3\frac{3}{4}$-ips four-track and eight-track "cartridge" tape machines (which are strictly for playing pre-recorded tapes). It is to be hoped that, ultimately, a breakthrough in technology will enable the system to reproduce frequencies up to 15,000 Hz and more, and that tapes themselves will be improved (by new methods of oxide application, etc.) to improve the marginally acceptable signal-to-noise ratios. After all, we can remember a time when everyone said that $7\frac{1}{2}$-inches-per-second machines would never be considered "hi-fi" because of their limited frequency response!

A Phono Question

Q. *What is meant by an "Anti-Skate" device?*

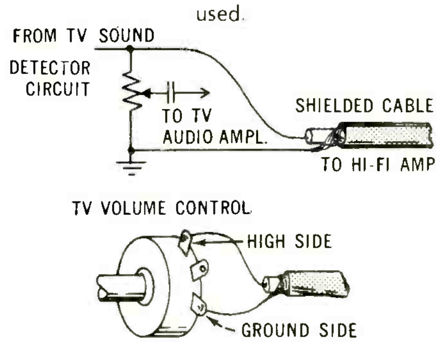
A. When a phono cartridge's stylus is playing a record, an inward force occurs due to the offset geometry of the tone arm. This side thrust can be observed by "playing" a blank vinyl disc—the tone arm will "skate" toward the center of the disc. Uncompensated for, this force causes some sound distortion, and tends to wear one of the sidewalls of the groove faster than the other. An anti-skate mechanism may consist of an additional spring mounted at the rear of the tone arm in such a way as to counter the inward force. An adjustable counterweight may also be utilized. A small refinement in record playing equipment, perhaps, but an important one that should not be overlooked.

General Questions

Q. *TV sound is so poor, as a rule. Why does everyone suggest connection of TV sound through a hi-fi system and how is this accomplished?*

A. TV sound, as recovered from the sound detector of a TV set, is surprisingly high in quality, having been transmitted by FM in much the same way as regular sound FM programs. What usually is poor in quality is the minimal amplifier and loudspeaker often built into TV sets by manufacturers. Most manufacturers concentrate on providing a good picture, spending little money or effort on the sound portion of the receiver. The best place to hook in electrically is at the volume control. As shown in Fig. 4, a shielded cable is used, with the inner conductor connected to the "high" side of the control and the shielded braid connected to chassis ground of the TV set. The other end of the cable is equipped with a phono-tip plug for connection to one of the unused inputs on your amplifier or receiver. You will be amazed at the improved quality of sound you will hear.

Fig. 4—Connection to TV volume control so that an external hi-fi amplifier may be used.



Using this connection method, the TV set's volume control should be turned down all the way so that no sound is heard from the TV set's own audio system. Connection to the speaker leads of the TV set is not nearly as good a method, for then you are involving the poor fidelity of the TV set's own audio amplifier. One

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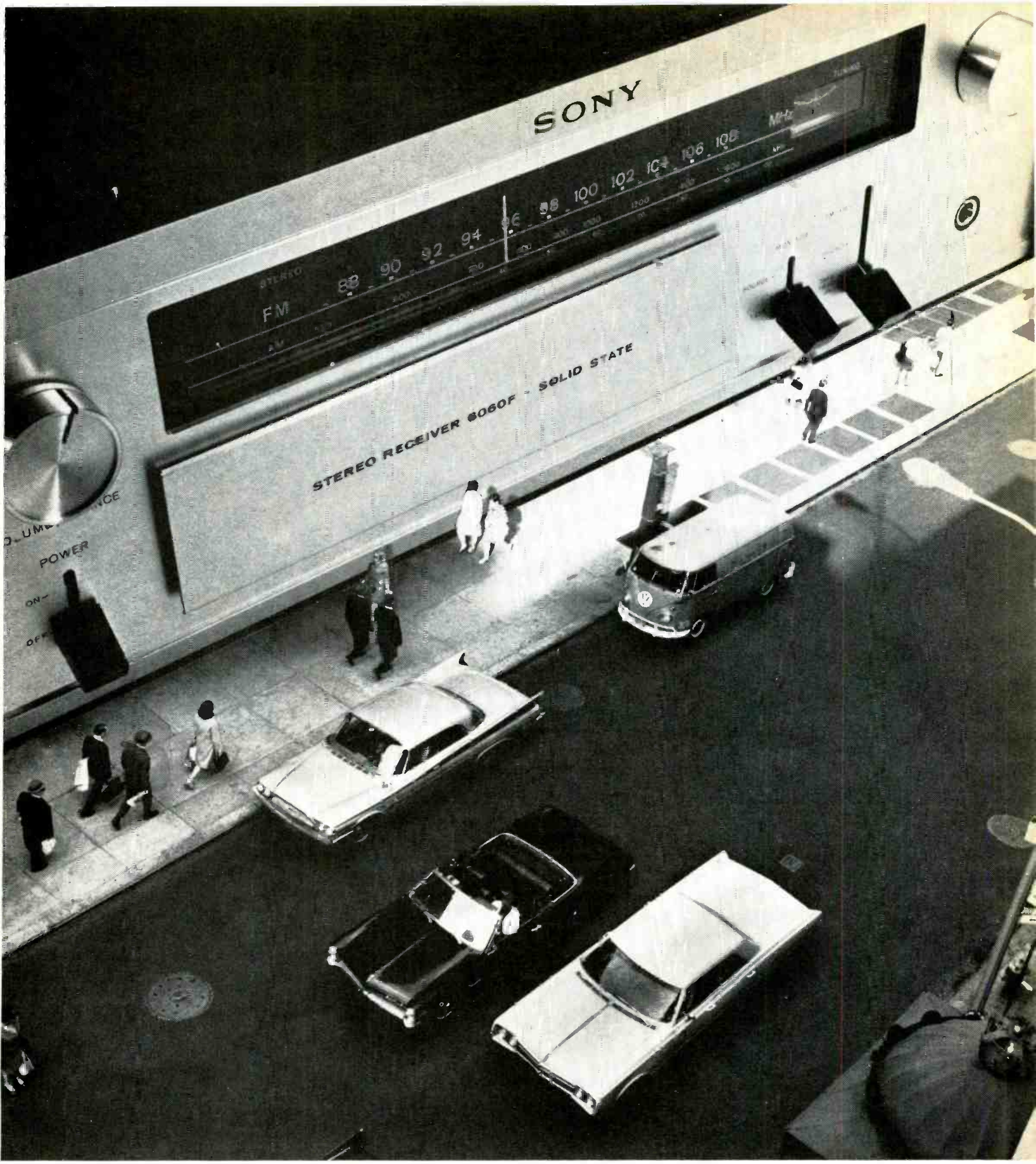

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The Sony®6060 receiver is the brightest thing that's happened to stereo hi-fi in a long while. A superb performer on FM stereo; FM and AM broadcasts; records and tapes. It will brighten up the music in your life.

Here's what Sony built: *Amplifier* – 110 watts IHF power into 8 ohms. Distortion less than 0.2% at rated output. *The tuner*—sensitivity 1.8uV. Exclusive solid-state i.f. filters never need alignment, provide razor-sharp selectivity, 80 dB; superb capture ratio, 1.5 dB. Spurious signal rejection, 90 dB. *Abundant control facilities:* automatic stereo reception; zero-center tuning meter; front panel headphone jack; switches for tape monitoring, muting, speaker selection, tape or Aux, input, loudness – the works.

At $399.50 (suggested list) the 6060 outshines receivers costing up to $500. Get a Sony disposition.

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Part 2
(Conclusion)

Altec Acousta-Voicing*

DON DAVIS

A practical discussion of how the acoustic properties of a room are actually "tuned"

Fig. 1. Professional sound contractors undergoing training in the set-up, calibration, and use of precision acoustical test equipment for performing Acousta-Voicing of sound systems.

THE DEVELOPMENT OF THE TECHNIQUES used to install and adjust sound system filters successfully is fascinating. To shape an electrical filter characteristic quickly, accurately, and economically when the inverse of the filter exactly matches the combined response of the transducers, electronics, and room requires skill. Difficult as this skill appears to be, experience has shown that most well established sound contractors do not find it hard to acquire.

The most difficult aspect of Acousta-Voicing has manifestly been the design of a sound system of adequate capabilities to support the increased performance possibilities the filters allow. Most often, failure to Acousta-Voice* a sound system successfully is not a fault of Acousta-Voicing filters and technique *but a failure to solve a basic sound system design or installation problem.* Any properly designed, carefully installed and rigorously tested sound system will respond to Acousta-Voicing and all its attendent benefits.

*T.M. LTV Ling Altec Inc.

If the sound system is inadequate, Acousta-Voicing oftentimes magnifies the problem. For example, take inadequate power availability. If the Acousta-Voicing succeeds in raising the overall system gain 10 dB and the power amplifier has only 5 dB more power available, then the system goes into total distortion.

Failure to establish even acoustical distribution is still another example. Before the Acousta-Voicing is performed, the difference in SPL between a position covered by the loudspeaker and a position receiving only the direct sound of the talker's voice may, if the acoustic gain figure is low, be as little as 2 or 3 dB. The observer in walking from the covered area to the uncovered area usually feels that it is down a little but he can get by. Then as Acousta-Voicing raises the acoustic gain by 15 to 20 dB, the same observer is startled to find a huge "hole" in the coverage. Acousta-Voicing won't cure design, engineering, or installation shortcomings.

What is adequate distribution? How is acoustic gain calculated? How much power is needed?[1] Where do the Acousta-Voicing equalizers go in the sound system, and what, if anything special, do they require of the system? These and many similar questions are the subject of the remainder of this article.

Typical Sound System Described

In order to describe Acousta-Voicing we will select a successful sound system as an example. It is outlined

[1] Arthur C. Davis and Don Davis, "Microphones for Sound Reinforcement Systems," AUDIO, December, 1967, p. 65.

in the single line diagram shown in Fig. 2. It is composed of the following carefully selected components:

"A" An Altec M-50 condenser cardioid microphone. This provides a microphone with smooth frequency response and a cardioid pattern, shock mounting facilities, wind protection, and stable long-term characteristics.

"B" An Altec 1588 microphone preamplifier. This unit provides the ability to handle high input levels at 150 ohms input impedance. It is a modular plug-in unit allowing simple service in case of malfunction.

"C" An Altec 1592 mixer amplifier. This unit accepts five inputs, any of which can be made microphone, phono, or line level, and can provide input impedances of 150, 600, or 47,000 ohms. The output power is sufficient to drive passive equalizer networks at levels high enough to stay above system noise.

"D" An Altec 15095 line matching transformer. This unit is used to convert the #5 input control on the mixer amplifier to 600 ohms "zero" level in order to accommodate the output of the wave analyzer.

"E" An Altec 9065A low-pass filter which provides attenuation of 3 dB at the selected frequency (usually 12,500 Hz) and increasing attenuation at the rate of 18 dB/octave *above* the selected frequency. This ensures that no response occurs above the bandpass capability of the filters being used to "tune" the system.

"F" An Altec 9066A high-pass filter which provides attenuation of 3 dB at the selected frequency (usually 70 Hz) and increasing attenuation at the rate of 18 dB/octave *below* the selected frequency. This ensures that no response occurs below the bandpass capabilities of the filters being used to "tune" the system.[2]

"G" Altec 9013-series narrow-band filters. These filters are tuned to each of the 24 standard ISO ⅓ octave band center frequencies from 63 to 12,500 Hz. The filters cross over with adjacent filters at their respective half-pad losses, allowing continuous amplitude response curves to be shaped. (See Fig. 3A.)

"H" Altec 9017-series broadband filters. These filters are tuned to each of the eight standard ISO octave band center frequencies from 63 Hz to 8000 Hz. These filters also cross over with

[2] Arthur C. Davis and Don Davis, "Professional Tone Controls," AUDIO, May, 1967, p. 60.

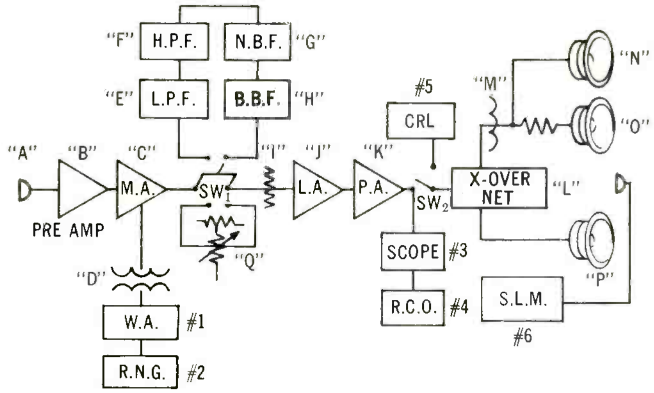


Fig. 2. Basic sound system layout for successful Acousta-Voicing results.

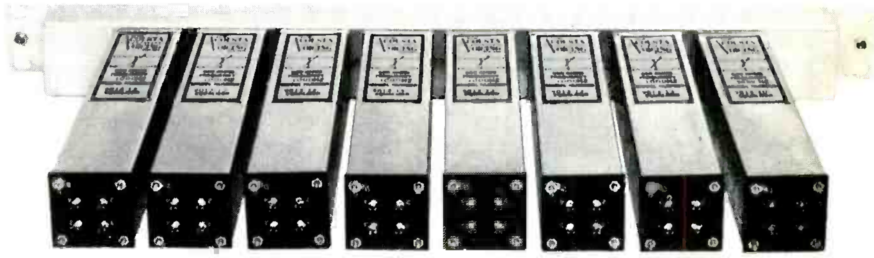
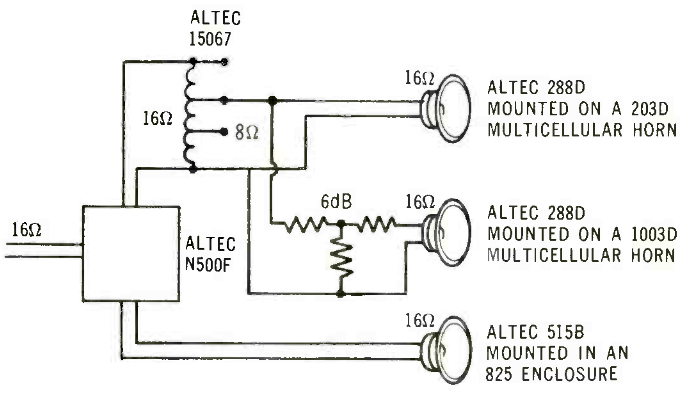


Fig. 3A.

Fig. 3B.

Fig. 3A. A series of 9013 narrowband filters mounted on the rack panel normally used in actual installations. **Fig. 3B.** A complete set of 9017 series broadband filters forming the 9018A filter set.

Fig. 4. This diagram shows how impedance matching (to ensure stability of the crossover frequency) and different power levels (to adjust for differences in level due to the inverse square law) are handled in an Acousta-Voicing system.



adjacent filters at their respective half-pad losses, allowing continuous amplitude response curves to be shaped. (See Fig. 3B.)

"I" A termination resistor. The value of this resistor is such that when it is placed in parallel with the input impedance of the line amplifier it brings the total impedance to 600 ohms.

"J" An Altec 1591A line amplifier. This unit provides 40 dB of gain capability allowing the restoration of the insertion loss of the filters.

"K" A 100-watt Altec 1594A power amplifier. This power is rationalized in the following manner:

1. Maximum acoustic level desired—85 dB SPL.
2. Maximum distance listener is from loudspeaker—150 feet.
3. Loudspeaker efficiency rating—113 dB SPL at 4 feet. There is a 32-dB drop in 150′ due to inverse square law; $113 - 32 = 81$ dB at 150′ at a 1-watt electrical input.
4. A 4-dB increase in power $= 2.5 \times 1$ watt, or 2.5 watts electrical input.
5. A peak ratio must be allowed to avoid distortion on all program material. A minimum (10 dB) is chosen, so 25 watts is required for the horn covering the rear of the church.
6. The horn covering the front of the church will require 6 dB less (since it has half the distance to cover), $25/4 = 6.25$ watts. *But it will require 25 watts input to the 6-dB pad preceding it.*
7. The woofer is 3 dB less efficient than the high-frequency units and will, after Acousta-Voicing, require 50 watts.
8. Totals are 25 watts to rear area, 25 watts to front area, and 50 watts to woofer, for a grand total of 100 watts. The power amplifier selected will deliver full power into a 16-ohm load from an input signal of −14 dBm.

"L" The crossover network is an Altec N-500F unit rated for use at 16 ohms (if properly terminated). Figure 4 shows the impedances and the connections of drivers to this network and their resultant a.c. impedance at the input to the network as measured with an impedance bridge. Unless proper impedance matching is maintained, the crossover frequency will shift with resulting undesired interaction with Acousta-Voicing filters.

"M" The Altec 15067 wide band-pass auto transformer, which allows changes in impedance from 2 to 1, 4 to 1, 1 to 4, and 1 to 2.

"N" The Altec 288D high-efficiency compression driver mounted on an Altec 203B two-cell multi-cellular horn and aimed at the rear audience area. One must be able to look through the throat of the mounted horn and "see" all seats it is intended to cover. Conversely, any seat covered by the horn must have an unobstructed view into one of the two cells.

"O" The Altec 288D mounted on an Altec 1003B ten-cell multicellular horn and aimed at the front half of the audience. Again, all seats to be covered are visible through the horn and can look back at the horn and see at least one cell.

"P" The Altec 515B completes the sound system. This is a 15″ low-frequency driver housed in an 825 phase-inversion enclosure that is horn loaded at the crossover region.

"Q" A bridged-T 600-ohm attenuator with 45 steps of 1 dB each, used in conjunction with Sw_1. This allows an instant switch between equalized and unequalized conditions with the attenuator replacing the insertion loss of the filters.

Testing of the Sound System Prior to Equalization

A beautifully designed sound system can be a dismal failure if not properly installed. From this problem stems the need for sound-system testing. Without converting this article into a book on testing[3] let's take a quick look at the test equipment used by over 100 Altec Lansing Acousta-Voicing contractors in the United States and Canada:

Equipment Required by Altec

1. General Radio Co., 1564A ⅓- and 1⁄10-octave bandpass Wave Analyzer *(WA)*. This unit not only serves as a tunable noise source when connected to the random noise generator but can be used as a precision voltmeter and distortion analyzer.

2. General Radio Co., 1383A Random Noise Generator *(RNG)*. It supplies "white noise" for loudspeaker distribution tests, "pink noise" for frequency-response tests when the band-pass filter analyzer is used, and USASI noise for measurement of acoustic gain.

3. Tektronix 504 Oscilloscope. It is used to trace hum and find oscillations in the sound system as well as to identify feedback frequencies and analyze transient conditions while tuning the system.

4. General Radio Co., 1309A RC Audio Oscillator. The oscillator is used in adjusting gain levels, for response tests of electronics, and for frequency identification of feedback frequencies.

5. General Radio Co., 1650A Impedance Bridge. The CRL impedance bridge is used for all a.c. impedance measurements in link circuits, loudspeaker lines, and so on.

6. General Radio 1565-9002 Sound Level Meter *(SLM)*. It is made only for Acousta-Voicing sound contractors under franchise to Altec Lansing. It features specially selected microphones and adjusted electronics, and the inclusion of a 4000 Hz octave band filter in place of its normal "A" scale.

These test instruments plus the master equalizer test set are used to perform Acousta-Voicing. Many contractors, in addition to these instruments, add a rms a.c.-VTVM, gain set, level recorder, tone-burst generator, oscilloscope camera, power-resistor decade, and frequency counter.

[3] Don Davis, *Acoustical Tests and Measurements*, Howard W. Sams, Indianapolis, ADH-1, 1965.

Adjusting the Broadband Acoustic Response of the Sound System

After all system tests are performed successfully and it is known that the loudspeakers are covering all areas evenly with sufficient power available, all impedances matched, and levels adjusted, we can proceed to the first step in the acoustic equalization.

The *RNG* is set to "pink" noise output because the ⅓ octave *WA* has *constant percentage* bandwidth and requires an input signal that drops 3 dB/octave with increasing frequency if its own output is to remain *constant amplitude*. The *RNG* output is connected to the *WA* input. The *WA* output is in turn connected to input 5 on the mixer amplifier through the 15095 matching transformer. Sw_1 is thrown to the filter position and the gain controls on the mixer, line amplifier, and power amplifier are adjusted for undistorted output from the power amplifier as observed on the oscilloscope.

The *WA* is tuned to the 1000-Hz ⅓-octave band and the acoustic output of the sound system is raised

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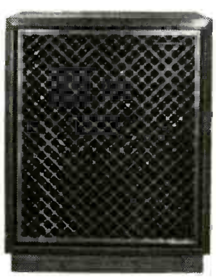

The sleek, contemporary Imperial I has a smart, walnut cabinet with a hand-rubbed French lacquer finish and is priced at $299.00. The elegant Imperial II, hand-crafted from selected hardwoods and finished in distressed antique, features a stunning hand-carved wood grille. It's yours for $369.00. Both possess a beauty of cabinetry equalled only by the beauty of their sound.

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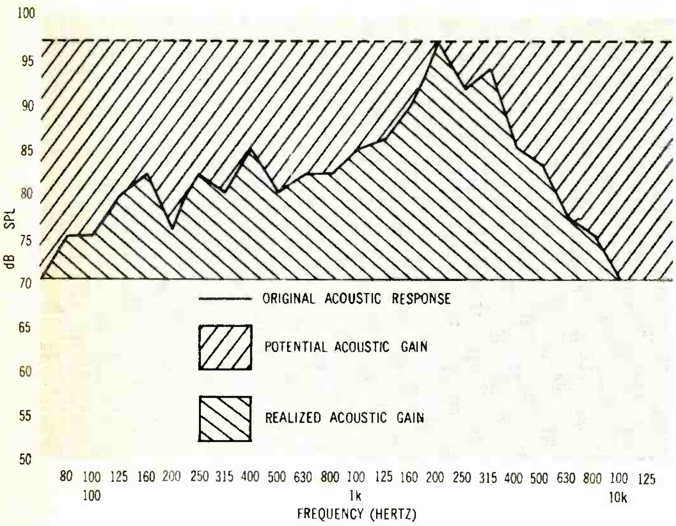

Fig. 5. A hypothetical curve of original acoustic response is illustrated. It shows the potential acoustic gain available, but not realized.

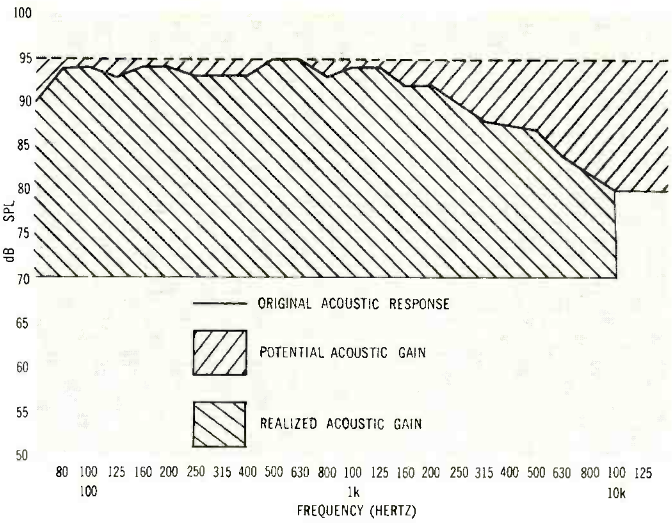

Fig. 6. The same system after Acousta-Voicing. The gain potential was realized by the control exerted through use of Acousta-Voicing filters.

until a reading on the *SLM* out in the audience area is obtained that is at least 20 dB above the ambient noise level. The *WA* is then tuned to the ⅓-octave band centered on 63 Hz and a reading is made on the *SLM*. Sufficient time must be allowed and enough samples taken to ensure a good average of the area at that frequency is obtained. This reading is then recorded on the chart shown in Fig. 5. Each subsequent ⅓-octave center frequency from 63 Hz to 12,500 Hz is read and plotted.

The curve on the chart in Fig. 5 is a hypothetical set of readings to illustrate the gain potential of a raw house curve. After obtaining the entire set of readings and plotting the curve, the Acousta-Voicing engineer applies broadband correction in octave-band sections to pull down the larger out-of-balance portion of the curve. Many subsequent runnings of this type of curve are performed, gradually adding first the broadband and then the narrowband filters to the circuit until the curve looks like that shown in Fig. 6.

The attenuator at "Q" in Fig. 2 is turned to maximum attenuation. With Sw_1 in the filter position, the system is brought up in level to the point where feedback is just beginning and then Sw_1 is thrown to the attenuator position. The attenuator is slowly turned toward less attenuation until feedback again just starts. The master gain control on the mixer is reduced and Sw_1 may be used to compare the difference in gain and quality between "filters in" and "filters out."

At this point in the tuning where the response curve looks like Fig. 6, the quality will sound excellent but the gain will not necessarily be high. The reason for this is that this curve represents the coupling of the loudspeaker and the room but it does not measure the coupling of the room and microphone. To solve this problem, a second technique is introduced as soon as the broadband response has been brought within the desired tolerances.

Control of System Feedback Frequencies

Calculation of the frequencies that are most likely to satisfy the Nyquist Criteria[4] of sufficient gain and correct phase angle staggers the imagination of anyone familiar with the complex interaction of architectural spaces with electro-acoustic systems. Fortunately, such a task need not be undertaken because of the "computer-like" attitude of the room in always selecting only one peak frequency at a time even when a great many lie only tenths of a dB away.

The procedure followed is simple. The gain of the sound system is raised until it feeds back. A little care with the control allows this feedback frequency to become a steady tone.

The vertical amplifier input of the oscilloscope is across the power amplifier output and the audio oscillator is connected to the horizontal input of the oscilloscope. This allows the oscillator to be used to "zero beat" the feedback frequency by means of a Lissajous figure. An exact "zero beat" is not essential but care should be taken to be sure that it is indeed the fundamental. While it is not at all difficult to train one's ear to detect "zero beat" audibly, it is not a good habit to get into. First, there is always the danger of mistaking a harmonic for the fundamental, especially at frequencies above 5000 Hz. Secondly, the oscilloscope reveals other troubles that may be developing such as instability, ringing, and so on. Finally, the oscilloscope allows instant detection of the exact insertion loss in the appropriate filter where regeneration ceases.

Due to the reverberation time of a room, it is very easy to over-deepen

[4] Richard V. Waterhouse, "Theory of Howlback in Reverberant Rooms," *J. Acous. Soc. Am.*, 37, 921, May 1965.

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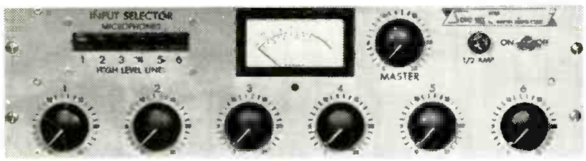

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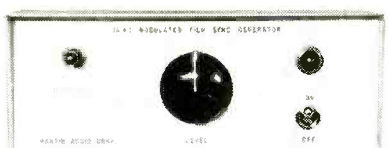

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the filter insertion loss when merely listened to and the Acousta-Voicer can find himself putting in 5- or 6-dB losses. When the oscilloscope is employed it is rare to exceed 3 dB and common to use 2 dB to control any single frequency. The Acousta-Voicer selects the filter frequency nearest the frequency on the oscillator dial and turns the attenuator one dB at a time until he observes the collapse of regeneration on the 'scope. (After "zero beating" with the Lissajous figure, the 'scope is switched back to vertical operation before inserting the filter in order to watch closely the effect of the filter's insertion on the waveform of the signal.) This procedure is repeated until the gain rises to the calculated maximum value potentially available with the microphone-loudspeaker spacing used.

Proximity Effects, Vibration, & Microphones

When microphones are approached by a body which is large compared to their own size, the sound field around them is upset and the system tends to sound as though it is ringing. This can be eliminated by using your hands around the microphone to force the system into feedback, and then using the appropriate filter to attenuate it. Normally, two of these filters are more than sufficient to eliminate any undesired change in the stability of the system when it is approached by an entertainer.

When a feedback frequency comes in quite strongly and persists after more than 5- or 6-dB loss has been inserted, it usually means that a microphone is picking up a resonance from some surface on which it is mounted. The solution, of course, is to remove the filter and isolate the microphone correctly, either through shock mounting or, if necessary, relocate to a firmer surface.

Final System Adjustments

Normally, at the end of the tuning, a high-pass filter at approximately 70 Hz is installed to help avoid low-frequency overload which can occur due to the greatly increased acoustic gain of the system and its increased susceptibility to footfalls on wooden stages, and so on.

A program equalizer is also normally installed. Inasmuch as the system may need to be brighter, for example, for one use than for another, the program equalizer allows the operator great freedom of final response.

The final test of this part of the tuning process is to run a detailed electrical response curve of the filters and, using 1000 Hz as a common reference point, compare the inverse of this curve with the original acoustic response curve. In this way it can be determined accurately if the filters have properly matched the raw house curve.

As can be imagined, it is wise to have the operating personnel who will be associated with the system observe the tuning. This can help to preclude later casual readjustment of power amplifier gains, filter settings, and so on, due to lack of knowledge of what is required to set them correctly. Because all Acousta-Voicing filters are calibrated and operate in 1-dB steps on a detented control, if someone accidentally detunes the system, it is a simple matter to return to the original settings. (Several copies of the settings are usually left on location with a master copy retained by the Acousta-Voicer.)

In a multiple-use space, a single set of filters can provide the necessary equalization for different sound systems. Once each system has been tuned and settings recorded, it requires only one minute of the sound operator's time to dial in the settings for a particular performance. This re-settability and ease of tuning means that Acousta-Voicing is particularly adaptable for rental and one-night tour productions. Also, due to the hour or so that is needed for each individual Acousta-Voicing, it is possible to tune a church, auditorium, or the like, for differing size audiences, since the size of the audience influences the house curve slightly.

Only those filters actually attenuated are left on location, usually mounted behind blank panels out of the way of curious hands.

Summary

Acousta-Voicing is an exciting development in the continuing progress of professional audio. New developments in its application are being found daily. To be successful in results, stable in operation, and versatile in application, the following minimum standards are most important:

1. The sound system must be well designed.
2. The sound system must be carefully installed and tested.
3. Accurate precision test equipment must be used to ensure accurate precision results.
4. Engineers engaged in Acousta-Voicing need rigorous training.
5. The filters must present a constant impedance to both source and load (especially at the frequencies of maximum insertion loss) to avoid serious transient response problems.
6. Filter depths of 28 dB must be available as the problems encountered can be of this magnitude.
7. Filters should be module. Otherwise the entire set of filters would have to be left in each installation. Step-type attenuators calibrated to 1 dB steps ensure accurate initial tuning and immediate resettability for dual usage.
8. Filter bandwidth must not be less than that of the critical bandwidth for the center frequency involved or ringing will occur.

The author would like to caution that, while this article is complete insofar as it goes, time and space have not allowed a discussion of all the pitfalls awaiting the unwary in attempting to carry out such a complex process. The whole subject of attempts to equalize with bandpass filters, and their unhappy consequences and other essentially negative hazards have only been alluded to. Acousta-Voicing is a tool of the professional. For those engineers interested in pursuing the subject further, a very careful perusal of the footnote material is strongly recommended. This, followed by the experience of witnessing an actual Acousta-Voicing session, followed by second and more careful reading of the footnote material, would be rewarding. Æ

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Equipment Profiles

This Month:

- Sony STR-6060FW AM/FM Stereo Receiver
- Sony TA-2000 Stereo Preamplifier
- TEAC A-4010S Stereo Tape Deck
- Stanton 681EE Stereo Cartridge
- JBL SE400S Stereo "Energizer"

(Also see Bose 901 Speaker Systems, page 10)

Sony Model STR-6060 FW AM/FM Stereo Receiver

MANUFACTURER'S SPECIFICATIONS—FM Tuner Section—IHF Usable Sensitivity: 1.8 μV. S/N Ratio: 65 dB. Capture Ratio: 1.5 dB. IHF Selectivity: 80 dB. Antenna: 300 ohm & 75 ohm. Frequency Response: 20 to 20,000 Hz ±1 dB. Image Rejection: 80 dB. IF Rejection: 90 dB. Spurious Rejection: 90 dB. AM Suppression: 50 dB. Total Harmonic Distortion: Mono, 0.3%; Stereo, 0.5%. FM Stereo Separation: More than 40 dB @ 1 kHz. AM Tuner Section—Sensitivity: 160 μV (built-in antenna); 10 μV (external antenna). S/N Ratio: 50 dB @ 5 mV input. Amplifier Section—Dynamic Power Output: 110 watts (total), 8 ohms. RMS Power Output: 45 watts per channel, 8 ohms. THD: Under 0.2% at rated output; under 0.08% at 0.5 watt output. IM: Under 0.2% at rated output; under 0.15% at 0.5 watts output. Frequency Response: Aux, Tape: 20 Hz to 60 kHz +0, −3 dB. S/N Ratio: Aux, Tape: 100 dB; Phono: 70 dB; Tape Head; 60 dB. Tone Control Range: Bass: ±10 dB @ 100 Hz; Treble: ±10 dB @ 10 kHz. General—Dimensions: 17 3/8 in. W x 5-5/16 in. H x 13-13/16 in. D. Price: $399.95.

This new receiver from Sony has a clean, smart-looking appearance. Perhaps it's the sight of a gold-anodized front panel that has only two metal-turned gold knobs plus three neat lever-activated switches. Maybe it's because the expansive dial scale has calibration marks at every MHz on FM, and they're all *equally spaced!* The impression is confirmed permanently as one starts to use this receiver for, despite outward panel simplicity, it has everything one could ask for in an integrated AM/FM-stereo receiver. It all gets back to "functional design" again. The secret of this layout is the hinged "trap door" which, when opened, discloses the properly arranged additional controls that you might have thought were missing at first glance. But let's not get ahead of ourselves.

To begin with, Fig. 1 shows the front panel layout, including a master volume control, tandem mounted with a fully separate balance control. Wisely, the balance control is operated by a small lever mounted behind the handsome volume control knob. Instead of the usual 300 degrees of rotation, it can only be moved to either side of center for about 45 degrees. When you think of it, most conventional balance controls don't do much for the first quarter of rotation in each direction anyway, so why build in the wasted motion? The huge dial scale is complemented on the right hand side by a tuning knob fully as massive and "comfortable" as the volume control at the left. The flywheel action is superb! The three levers along the lower section of the panel are used for "power on/off," "tape monitor" and a three-position "selector" switch. Yes, we said *three*—just FM/AM, Phono, and Tape Head/Aux.

How can you have a position that reads AM and FM at the same time? Easy, if you examine what's behind that little "closed door," as shown in Fig. 2. Starting at the left are two little lever switches (miniature replicas of their bigger mates on the front panel proper). They turn on (or off) either the main or remote speakers. Next come the expected bass and treble controls (ganged, in each case). The not-so-often-used loudness contour switch (another lever) comes next, followed by levers for mono-stereo automatic (in FM), choice of tape-head *or* aux inputs and choice of muting, in or out. Finally, there is a two-position, small rotary switch knob which selects AM or FM. This "switching upon switching" arrangement lets you set up the receiver to favor your most-often-used services behind the trap door and then allows you to select them by means of the simple, three-position main lever selector switch.

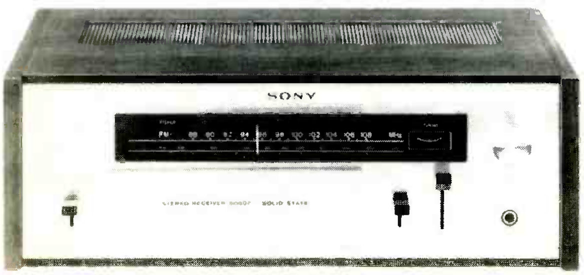

Fig. 1—Sony STR-6060FW FM stereo/FM/AM solid-state receiver.

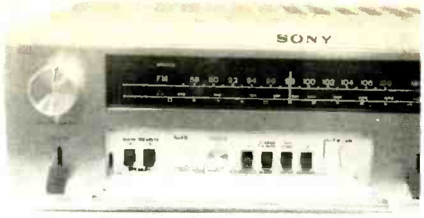

Fig. 2

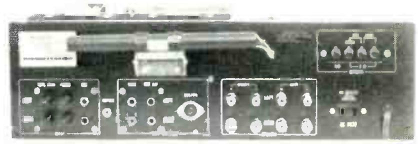

Fig. 3

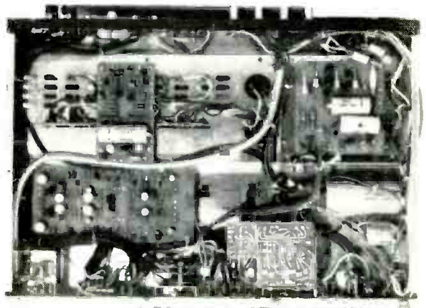

Fig. 4

Fig. 2 (top)—A "trapdoor" hides less-frequently-used controls, providing a less-cluttered appearance.

Fig. 3—Rear panel layout.

Fig. 4—Chassis (with bottom cover removed) discloses a neat, modular layout and husky components.

The dial glass area also contains the usual stereo indicator light, plus individual lights denoting AM or FM operation. To the right of the numerals is a dual-purpose tuning meter: zero centered for FM, and perfectly calibrated (we checked its centering against a signal generator, using sweep alignment techniques), and "peak-reading" for AM. Few receivers we have seen provide tuning meter operation for AM as well as for FM.

The rear panel contains the requisite number of inputs and outputs, a useful ground terminal, a special jack for imported tape recorders, one convenience a.c. outlet (unswitched), and antenna terminals for 75- and 300-ohm connection in FM as well as provision for an external AM antenna to supplement the AM adjustable loopstick antenna that comes as part of the receiver. Since the output stages include protective circuitry that is self resetting, there is no need for external replaceable fuses. Rear panel layout is shown in Fig. 3.

While the front panel may look simple, the circuit is as advanced as any we have seen. The FM front end uses 2 field-effect transistors (FET's) and a five-gang variable capacitor, which accounts in part for the high sensitivity, extremely high overload capacity and excellent cross-modulation characteristics noted during measurement.

Eighty dB of FM selectivity is achieved, thanks to newly designed crystal i.f. filters which replace conventional tuned transformers in the i.f. section. We found that the muting circuit silenced interstation noise (as well as signals under 5 μV, as specified) while contributing *no* increase in distortion for slightly stronger signals which were able to "break through" the muting feature.

Circuit layout is excellent; major parts examined seem quite rugged and of good quality. An underneath look at the chassis, including several of the printed circuit modules, can be seen in Fig. 4.

Performance

With receivers getting more and more sensitive, we have started a new procedure in "counting" stations received. We now use both the indoor dipole (usually supplied by the manufacturer) and our regular outdoor FM antenna, and state results for both. In this case we received 37 acceptable signals (12 stereo) with just the dipole, while the outdoor antenna raised the total to 39 (13 stereo).

This came as no great surprise, for we had already examined sensitivity and other FM traits, as plotted in Fig. 5. Our plot shows an average sensitivity of 2.0 μV against the claimed 1.8, which is really splitting hairs, since we *did* measure 1.8 μV at 88 MHz and 2.1 at 106 MHz. This variation is negligible, as production tolerances go. S/N measured 65 dB as claimed, while THD was 0.3% in mono and only 0.4% in stereo. Stereo separation for FM measured 40 dB at 1 kHz, with figures at other frequencies plotted in Fig. 6. While Sony claims separation of "better than 40 dB" at 1 kHz, you couldn't prove it by us, for our equipment is only capable of measuring down to 40 dB!

If Sony wants to rate the power output of each channel at 45 watts (referenced to 0.2% THD at 8 ohms) that's very nice of them, but how conservative can you get? We can report that they do reach 0.2% THD at *exactly* 45 watts per channel, as shown in Fig. 7, but look beyond 45 watts and you find that they don't reach 1% THD per channel until 55 watts per channel! In the case of IM, while it also reaches 0.2% at 45 watts, it's only up to 0.38% at 50 watts per channel!

While Sony doesn't mention power bandwidth, we found it to extend from 20 Hz to 45 kHz, as shown in Fig. 8, while frequency response was within 1.5 dB from 20 Hz to 60 kHz—again better than published claims. Tone control range is just about average, as shown in Fig. 9.

Using the Sony STR-6060 FW receiver for a couple of weeks was a very pleasant experience. While record reproduction was as good as any we had heard, there's something about the FM action that's just a bit better than most. Maybe it's those crystal filters—but whatever it is, FM sounded more "live" than usual—imparting a very crisp, natural-sounding equalization to several of the better stations in our area that we seldom hear quite that good. Even the AM section, which we don't usually measure with instrumentation, seems particularly "wide-band" and almost of "high fidelity" quality. Such excellent receiver performance at a $400 list price is not so very common these days.

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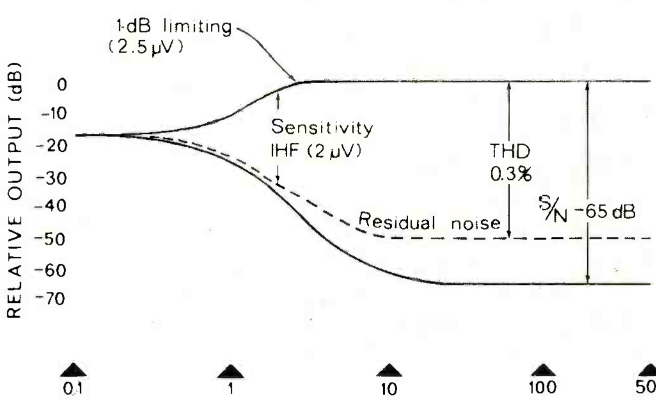


Fig. 5—FM characteristics of the Sony STR-6060FW receiver.

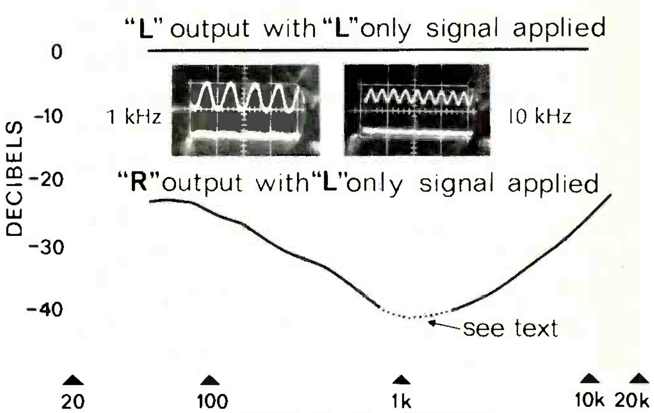


Fig. 6—Stereo FM separation measurements. Dual 'scope traces shown at 1 kHz and 10 kHz.

Fig. 7—THD and IM distortion characteristics.

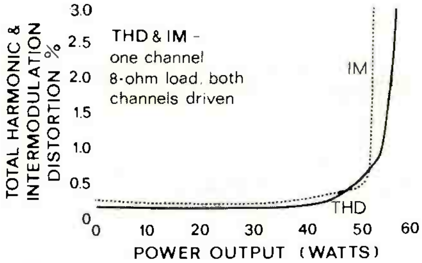


Fig. 8—Power bandwidth, based on 0.5% THD and 50 watts.

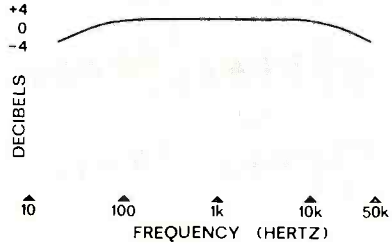
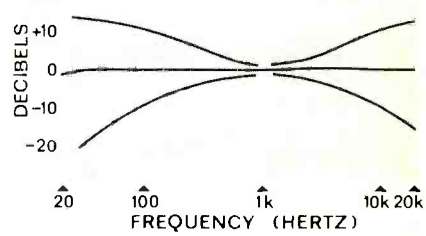


Fig. 9—Bass and treble tone-control range.



Sony Stereo Preamplifier, Model TA-2000

MANUFACTURER'S SPECIFICATIONS— Input Sensitivities—Tape Head, Phono 1, Phono 2 (Normal): 1.2 mV max. (adjustable. Phono 2 (low): 0.06 mV max. (adjustable). Tuner, Aux-1, Aux-2, Tape in, Rec/PB: 120 mV (all adjustable) max. Microphone: 1.2 mV. Max. Low-Level Inputs: 100 mV. Rated Output Levels—Rec. Out: 120 mV into 1.5k ohms, max. 10 V. Preamp Out, Line Out: Choice, 0.3 V or 1 V., 10K ohms (max. 2.5 V). Center Channel Out: Choice, 0.45 V or 1.5 V, 7.5K ohms (max. 5 V). Headphone Out: 3 V, 1.5K ohms (max. 10 V). REC/PB (Output): 120 mV, 80k ohms (max. 1 V). Harmonic Distortion, 1 kHz @ rated output—High-Level Inputs: Under 0.03%. Low-Level Inputs: Under 0.05%. IM Distortion @ rated output—High-Level Sources: Under 0.05%. Low-Level Inputs: Under 0.07%. Frequency Response — Tuner, Aux-1, Aux-2, Tape In: 20 Hz to 100 kHz +0, −2 dB. Phono 1, Phono 2: RIAA within ±0.5 dB. Tape Head: NAB within ±0.5 dB (adjustable). Microphone: 20 Hz to 30 kHz, +0, −2 dB. Signal-to-Noise Ratios—High Level Inputs, 200 mV applied: 90 dB. Phonos 1 & 2, 3 mV applied: 70 dB. Phono 2, low (0.1 mV applied): 50 dB. Tape Head, 1.5 mV applied: 65 dB. Microphone, 1.5 mV applied: 65 dB. Tone Control Ranges—Bass (in 2-dB steps, total of 11 steps): ±10 dB @ 100 Hz. Treble (in 2-dB steps, total 11 steps): ±10 dB @ 10 kHz. Filter Actions—High Filter: 12 dB/octave cut above 9 kHz. Low Filter: 12 dB/octave cut below 50 Hz. General — A.C. Convenience Outlets: 3 switched, 1 unswitched. Dimensions: 15¾ in. W x 5½ in. H x 12⅜ in. D. Weight: 19 lbs. 4 oz. Accessories supplied: 6 shorting plugs, 4 pin plugs, 4 patch cords. Price: $329.50.

Not intended for "mass consumption" by any means, the Sony Stereo Preamplifier, Model TA-2000 Preamplifier Control Chassis, at a suggested list price of $329.00, is "professional" in both appearance and performance. For old audiophiles such as ourselves it is nice to know that there is now a growing choice of such solid-state chassis on the market again. You may be sure that there will always be room for such preamp-control units, for there will always be those who insist upon the utmost in control flexibility, together with being able to choose from among the better separate power amplifiers (often hidden from sight), separate tuners, and other components. Judging by this design, it is sure to find its way into many a modestly equipped recording studio as well, for it deserves to be classed with professional studio equipment.

The front panel, shown in Fig. 1, is made of the same heavy, gold-anodized aluminum as the Sony receiver reviewed in the preceding Profile. But whereas the receiver has hidden controls, to give it a simple look, the TA-2000 is resplendent with all sorts of wonderful knobs, levers, jacks, and meters. Somehow, though, they all seem to "belong" on this panel; about five minutes of instruction-book scanning serves to explain them all.

In brief, there's a smooth-acting master volume control at the left, followed by left- and right-channel illuminated VU meters. These meters, by the way, have two sensitivity ranges, selected by means of a push-push switch. In the test position, 0 VU corresponds to "rated output" (either 0.3 or 1.0 volts, depending upon the sensitivity of your power amplifiers). In the normal position, an extra 14 dB of sensitivity is imparted to the meter movements, so that they are more responsive during actual use, which usually involves levels well below rated output. Step-type switch tone controls come next, and each step is worth 2 dB (at 100 Hz for the bass control and at 10 kHz for the treble control), for a maximum range of precisely ±10 dB. The mode switch permits listening to the left speaker or right speaker only, stereo and reverse (remember when all equipment had a "stereo reverse" position?), L + R mono and left or right channels to both speakers. At the upper right of the panel is a function-switch arrangement consisting of a three-position lever switch *plus* a five-position rotary switch. The design philosophy is again similar to that employed in the Sony 6060 receiver. The lever switch "up" and "down" positions select the most popular program sources—phono and tuner, while the mid-position swings selection over to the rotary switch for such less-often-used sources as Mic, Tape Head, Phono 2, Aux 1 and Aux 2.

The lower section of the panel includes a separate lever for power on/off, a "line" output jack (for connection to tape recorders without having to get at the back of the instrument), a stereo headphone jack with a very welcome headphone level control, lever switches for actuating low- and high-frequency cut-off filters (having a meaningful 12 dB/octave slope starting at 50 Hz and 9 kHz, respectively) flanking a center lever switch that cancels or bypasses the stepped tone control switches completely (for the real purist who wants everything absolutely FLAT). Aux 2 inputs are in the form of a front panel jack, making it easy to dub recordings from your friend's tape recorder, again without having to get around to the back of the set. Left and right microphone inputs are also located on the front panel for the same ease of accessibility. Finally, at the lower right, another lever switches the circuit in and out for tape monitoring.

The rear panel, in addition to the necessary input and output jacks, features individual input level adjustment controls for every single input jack pair, plus an output level switch for choosing 0.3 volts or 1.0 volt out, depending upon power amplifier sensitivity. The phono 2 input has a slide switch associated with it which offers a choice of input sensitivity. In the "low" position, it will accept any of the extremely low-output cartridges which would otherwise require a step-up transformer or pre-preamp. A mixed center channel output jack, together with its own output level control, a recorder jack designed to fit standard imported recorders and *four* (count

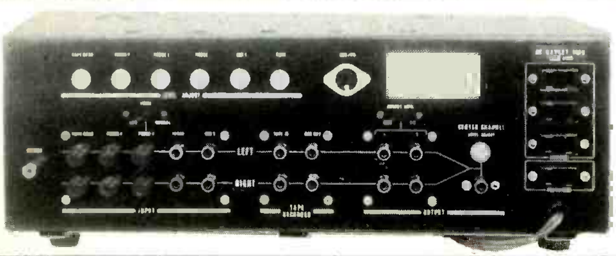
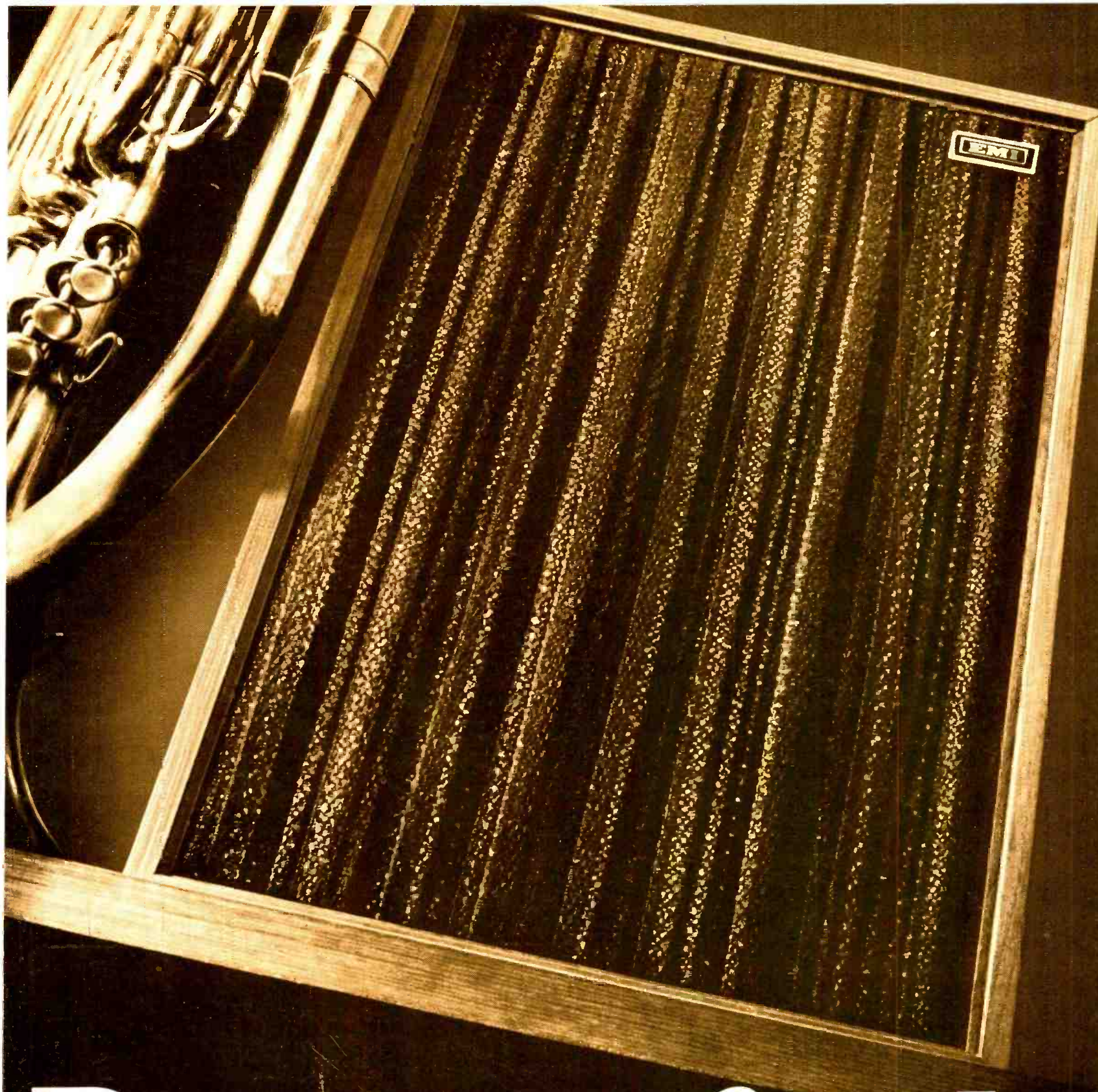


Fig. 2 — Back-panel layout of TA-2000 preamplifier.



Basso profundo

Basso Profundo... the richest, biggest bass voice in the world. That's just one characteristic the new EMI 205 has that other bookshelf speakers don't.

We squeezed more bass into the 205 by using an elliptically-shaped woofer rather than a round one. This gave us more sound-generating cone area than if we had used a round woofer. The bigger the cone area, the bigger the sound. And the woofer's rigid cast-frame construction eliminates spurious sounds and contributes to the 205's smooth bass response.

But there's more to the EMI 205 than just its uniform bass response. Specially damped tweeters provide smooth, transparent response into the highest frequencies. Three crossover networks with special switch controls for tweeter and mid-range let you tailor the response to your personal listening tastes and to the acoustics of your room.

The new EMI 205 reproduces the entire audio spectrum faithfully, flawlessly—even at low listening levels. In handsome, oil-finished walnut enclosure with dimensional pleated grille, $225. Choose from a complete line of EMI speaker systems starting at $54.95. Benjamin Electronic Sound Corp. Farmingdale, New York 11735 (available in Canada).

New EMI 205

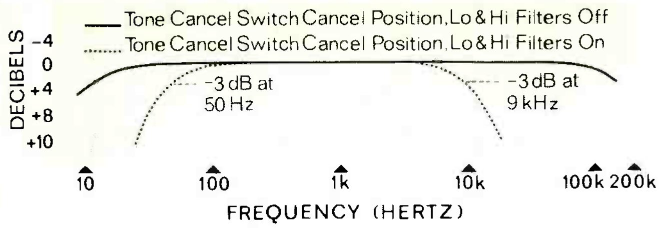


Fig. 3 — Frequency response with and without LOW and HIGH filters in the circuit.

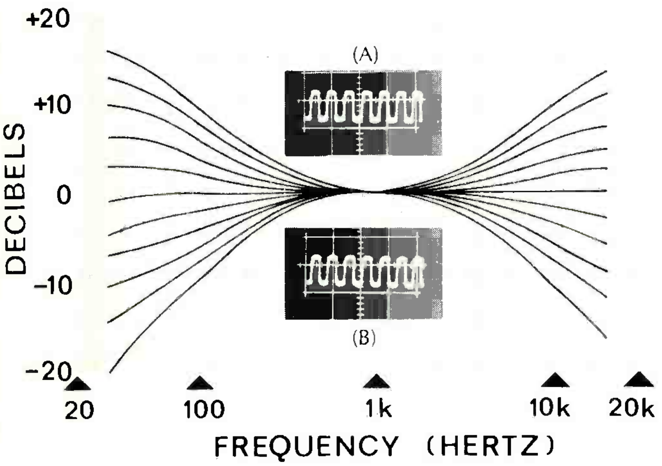


Fig. 4 — Step-type bass and treble controls provide precise curves. Center curve is "flat" setting, with tone controls active. Trace (A) shows 10-kHz square-wave response with tone controls bypassed. Trace (B) shows results with tone control "in-circuit." In flat position, some evidence of "rounding off" at waveform top is seen, though even with this setting uniform response was obtained up to 60 kHz.

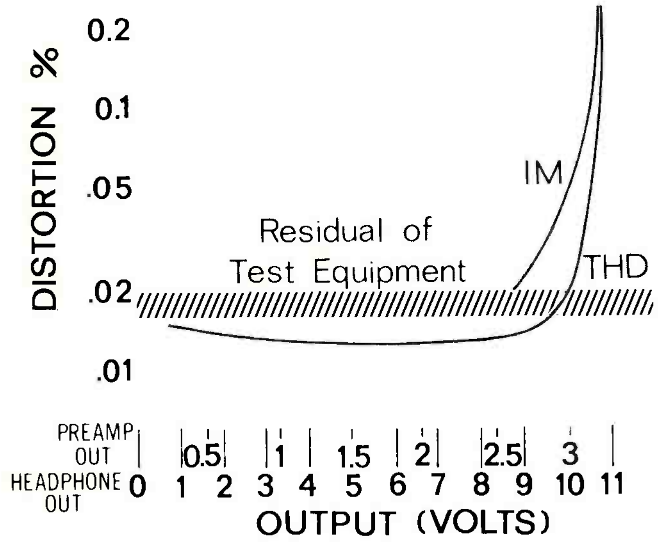


Fig. 5 — Distortion curves are shown, using manufacturer's drawings (see text).

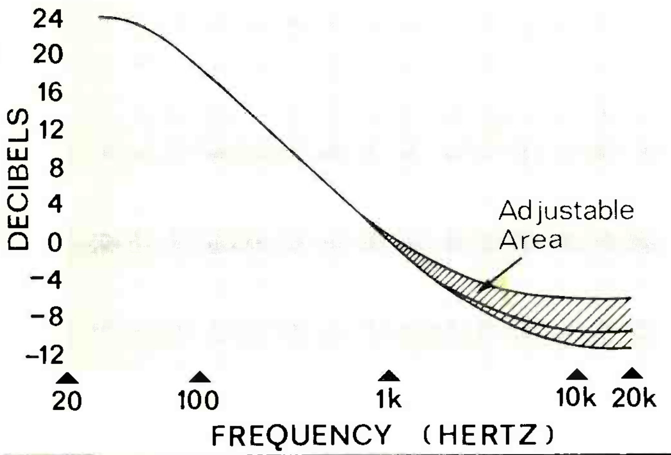


Fig. 6 — Tape-head equalization can be adjusted to compensate for different playback tape equipment.

'em) a.c. convenience outlets (one unswitched) complete the back panel layout. (See Fig. 2.)

We plotted frequency response (Fig. 3) and tone-control range (Fig. 4) for this fine unit, as well as the low- and high-frequency filter action (Fig. 3). We must admit, however, that when we tried to measure distortion at normal operating levels (both THD and IM), it was too low for our test equipment. Accordingly, we present these measurements as published by the manufacturer in their comprehensive instruction manual, as shown in Fig. 5.

We measured phono equalization, too, and found it was so precise that reproducing the curve would mean simply re-publishing the standard RIAA curve itself. When we first tried to repeat the procedure for tape-head equalization, however, we found the high end to be off by 2 dB—but that was before we discovered that the preamp has adjustable tape equalization to suit the needs of tape playback units that don't quite conform to NAB standards. Figure 6, then, shows the results obtained at the extremes of this adjustment (located under the chassis), as well as the standard NAB response curve.

What superlatives can be applied to the dynamic performance of this unit that haven't been used before? For one thing, the residual noise is so low that even magnetic cartridges are afforded a dynamic range of over 80 dB, referenced to 5 millivolts of input. High-level inputs are good for 90 dB of signal to noise (assuming the source is that good). This is about the limit of most good power amplifiers—where it's a lot easier to do. The frequency-response measurements were deliberately taken with the tone controls in the circuit (set to the "flat" step position, of course), and the response was excellent. Yet, when we took photos of 10-kHz square-wave response we found that the wave was "squarer" with the tone controls "cancelled out" than when the tone controls were active (Fig. 4), indicating that the tone controls, even when set flat, do alter this response (possibly because of some phase shifting), justifying even this last frill engineered by Sony.

Frankly, we're all for integrated receivers — they've brought stereo hi-fi components to the point of "almost mass acceptance." But when we encounter a unit such as the Sony TA-2000, we long for the old, "original approach"—even at $329.95!

TEAC has a sound effect on Christmas shoppers.

And this tape deck is a sound value, too. Maybe that's why so many Christmas shoppers are buying it for themselves. It's our standard four-track model, with all the famous TEAC quality at a happy holiday price. And plenty of unique features, like the popular ADD recording for simultaneous playback and recording on separate tracks. (Now you can harmonize with yourself.) Two exclusive eddy-current outer rotor motors drive the reels; a dual-speed hysteresis synchronous motor drives the capstan. And this machine drives our competitors crazy — especially since it was *recommended by the leading report to consumers.* There's a lot more happening under that handsome walnut styling, too . . .

• All-pushbutton controls — touch and go! • Automatic shutoff • Independent LINE and MIC input controls • Stereo echo for special sound effects
And a merry Christmas to you!

(Now you're really hearing things.)

TEAC

TEAC Corporation of America • 2000 Colorado Avenue • Santa Monica, California 90404

Equipment Profiles (continued)

TEAC Model A-4010S Stereo Tape Deck

MANUFACTURER'S SPECIFICATIONS— Tape speeds: 7½ & 3¾ ips. Motors: 2 outer-rotor types for reel drive; 1 hysteresis-synchronous for capstan drive. Reel size: 7" max. Heads: Four; 4-track stereo, erase, forward record, forward playback, and reverse playback. Wow & flutter: 0.12% at 7½ ips; 0.15% at 3¾ ips. Freq. Resp.: 30-20,000 Hz at 7½ (50-15,000 ±3 dB); 40-12,000 at 3¾ (50-7500 ±3 dB). S/N: 50 dB. Crosstalk: 50 dB channel-to-channel at 1000 Hz; 40 dB between adjacent tracks at 100 Hz. Inputs: Microphone, 10,000 ohms, 0.25 mV min.; Line, 100,000 ohms, 0.14 V min. Output, 1.0 V for load impedance of 10,000 ohms or more. Dimensions: 17⅜" W x 17-7/16" H x 8½" D. Weight: 44 lbs. Price: $469.50.

The TEAC A-4010S is a combination of the A-4000S tape player and the RA-40S recording amplifier. The player/transport is 11½ in. high, and the recording amplifier is 4 in. high. Both are mounted together in a single, attractive wooden cabinet to provide both recording and playback operation in a single unit. The machine is capable of recording and playing in the forward direction, but only of playing in the reverse direction. Reversing is automatic when a metallic tape is applied to the back of the tape at the point where reversing is desired.

The A-4010S is push-button operated, and the various actions are initiated by relays—seven in all—which provide the various functions. The front view, Fig. 1, shows the two units in place—the lower section being the recording amplifier. The playback amplifier is located under the right spooling motor, shown at the left of Fig. 2. When used as a tape player only, the output is at a fixed level, with speed equalization being controlled by the speed switch.

With the record amplifier being used, the two units are connected by two single shielded cables for the playback output, and by two cables consisting of six and eight leads respectively—the first provides for switching the record equalization, and the second connects to the record heads with the recording signal, and provides for bias and erase connections, as well as a.c. power to the amplifier. Figure 2 also shows the rear of the tape player section only. Also available on the back panel are: a socket for remote control which can be used to stop the tape motion; or to start it in either direction; a male receptacle for the a.c. line cord; the line-voltage selector and frequency selector switches, two fuses—one for the transport and one for the record amplifier; and a standard DIN socket for connecting to an amplifier so equipped. The use of this connection, however, eliminates the ability to mix high-level and microphone signals in recording.

The front panel of the recorder/transport, Fig. 1, accommodates the two reel hubs, each with ridged rubber rings to provide better contact with the reels. Above them at the center of the panel is a four-digit counter with a push-button reset, while below the reels and also centered is a switch button to control tape tension. In the depressed position, tape tension is decreased for use with ½-mil tapes. At the left are four rectangular push-button switches marked FAST and STOP. These buttons energize the necessary relays and solenoids for operation of the transport. Above them is a tape-tension lever, followed by the reversing contacts—a pair of metal rods imbedded in the tape guide. To their right is the tape-head cover, which houses the four heads, shown with the cover removed in Fig. 3, a photo of the entire unit with the front panel removed. Further to the right are the capstan, pressure roller, power switch and indicator light, and the two buttons for changing

Fig. 2—Three motors of the TEAC A-4010S deck are seen with the back panel removed.

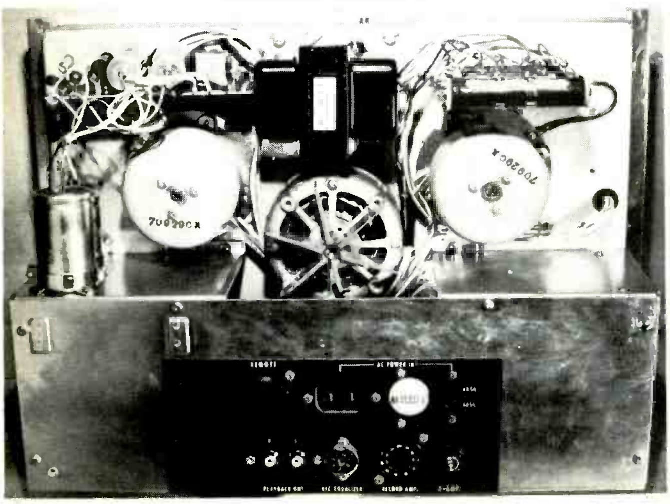
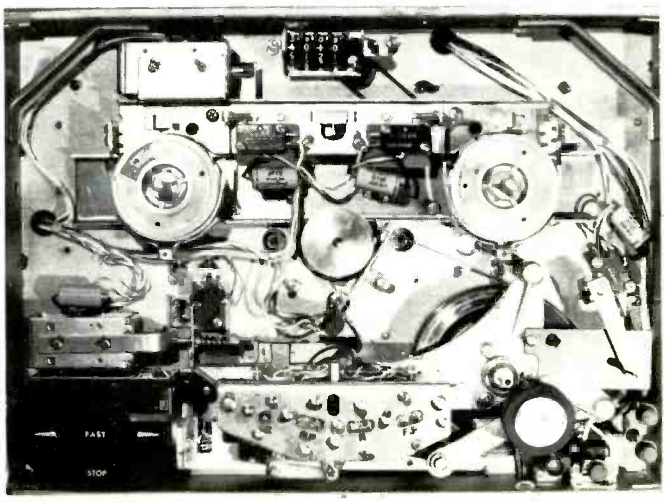


Fig. 3—Here is a view of the recorder with the top panel removed.



new star of the show!

Koss model esp-6 electrostatic stereophones*

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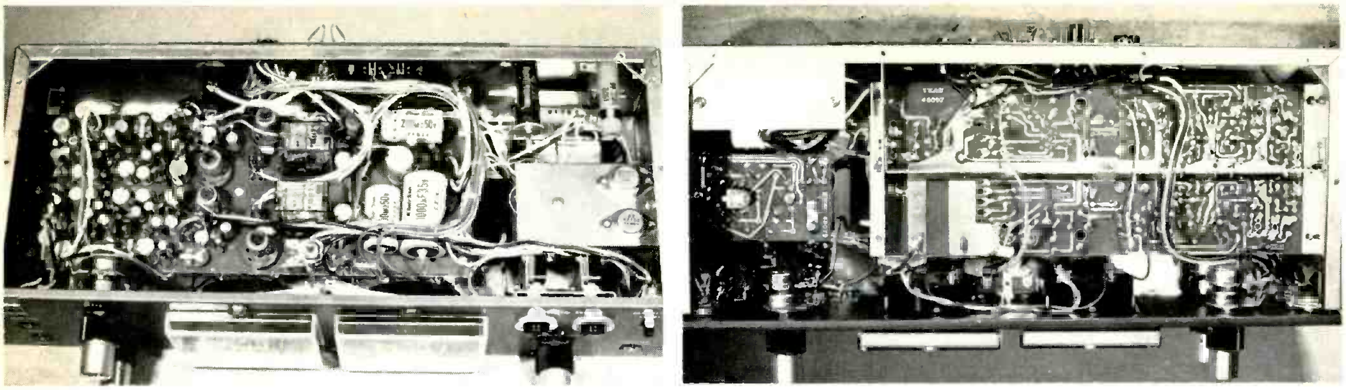


Fig. 4—Top and bottom views of the TEAC A-4010S's recording amplifier.

speeds. Above is the automatic shut-off lever, which actuates a microswitch to turn off power when tape runs out.

On the recording amplifier panel at the left are the two microphone input jacks, followed by two dual-concentric record-level controls, one for microphone input and one for line input. Next are the two big 4½-in. VU meters, with the record indicators above each. These are followed by two RECORD pushbuttons, one for each channel, the monitor switch to select monitoring and output from either source or tape, a dual-concentric playback level control, and a headphone jack.

Speed change operates by electrical switching of the windings in the capstan motor. Fifty to 60 cycle power lines can be accommodated by moving the drive belt (between motor and flywheel) from one step on the motor pulley to the other. Figure 4 shows the top and bottom views of the recording amplifier. The playback amplifier consists of two identical sections, each employing two transistor feedback pairs plus another transistor as a voltage regulator. In the recording amplifier, there are two 4-transistor amplifier sections for the record function, a push-pull bias/erase oscillator operating at 108 kHz (in our test sample), a solid-state regulator stage, the playback level controls, and two 2-transistor amplifiers which provide additional gain to drive the VU meters from an adjustable potentiometer in the record amplifiers, and to provide for switching monitoring and output from tape to source.

Operation

The A-4010S is an exceptionally easy machine to operate. It is easy to thread, and is apparently foolproof. It is difficult to move fast enough to cause the tape to break by improper operation of the transport buttons. The record interlock indicators are neon lamps which are fed with the output of the bias oscillator, so if the lights are on, you know that your bias oscillator is working. Microphone input jacks are terminated if no microphone is plugged in, and mixing of microphone and line is done after the microphone preamplifier stage. An adjustment is provided for balancing the oscillator for optimum symmetry (and consequent minimum of tape noise).

The transport is provided with adjustable wire-wound resistors — seem to be about 50-W. types—for setting up the proper voltages for the several functions of the reel motors. Functions are normal wind for recording, fast wind, and holdback—which for two-direction operation is a most desirable feature, and entirely unexpected in a machine designed for home use. All three motors have their series phase-shifting capacipacitors switched when the speed is changed, and all switches are shunted with click suppressors. On the whole, the machine is equipped with practically every convenience that could be wanted, even if it will not record in the reverse direction.

The weight of the machine should, by itself, indicate the sturdiness of construction and the completeness of the overall design. There are seven relays and two solenoids in the unit, together with 24 transistors and 8 diodes, with all the transistors being germanium PNP types except those in the bias oscillator circuit, which are NPNs.

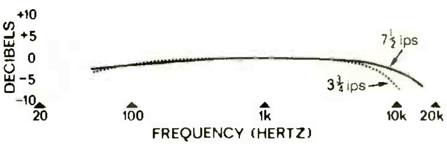
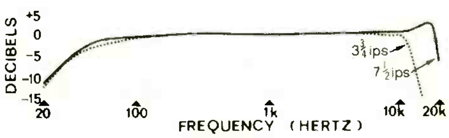


Fig. 5—TEAC playback response to standard recording tape.

Fig. 6 – Record/playback frequency response of the A-4010S deck.



Performance

In the performance category, the A-4010S passes with flying colors, exceeding its specifications appreciably. Frequency response at 7½ ips measured within ±3 dB from 35 to 19,000 Hz, using Scotch 202 as the recording medium, and without readjusting bias or disturbing the factory settings of equalization nor realigning the azimuth of the playback heads. At 3¾ ips, the frequency response extended from 32 to 12,000 Hz ±3 dB. Wow and flutter measured .07% at 7½ ips, and 0.11% at 3¾, both figures being better than specs. Winding time measured 70 secs. for a 1200-ft. reel, as against 90 secs. in the specs. Signal-to-noise was a creditable 56 dB at 7½, and 44 dB at 3¾ ips, and channel separation was 49 dB at 1000 Hz on both stereo and mono. Separation between adjacent tracks—important with a reverse-play machine—measured 47 dB. Record/play distortion measured 0.6% at −10 dB with respect to the 0 on the meters, and the 3% point occurred at a level 8 dB above the indicated 0 level on the meters. The frequency response curves for both playback from standard tape and the record/play response are shown in Figs. 5 and 6. The input required for a 0 level on mike input was measured at 0.1 mV, and for line input it was a modest .07 V, which is 6 dB below specifications.

On the whole, the TEAC A-4010S stacks up as a fine machine—one which is easy to handle, excellent in performance, and a unit which is able to complement any fine stereo hi-fi system. For those who want a 2-track recorder, such a model is available.

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THE CASE OF THE VANISHING TWEET

AND OTHER STEREO MYSTERIES

High frequencies are a sometime thing for most audiophiles.

Sit quietly where X marks the spot between your well-aimed tweeters. and every delicate overtone is audible. Stand up, move around, or even turn your head, and *pouf!* Those vital upper harmonics that give music its color and texture have simply disappeared.

The culprit is *directionality,* which every tweeter has unless you take steps to remove it.

We did. Completely.

We equipped our Grenadier speaker systems with a Divergent Acoustic Lens.

If you know your acoustical physics, or your musical math, you know that the higher a tone is, the more it 'beams' on a straight-line axis. When you get into those fine upper partials, moving your head an inch off-axis can give you vanishing tweet. *Unless* your speaker spreads them out.

We use a *domed* tweeter to spread them at the source. Then we use our Divergent Acoustic Lens to distribute them through a 140° arc.

Instead of high frequencies that beam out of a box like this ⊏→ , you get highs that radiate from our stereo cylinder like this

In other words, non-directionality.

And we don't leave it at that.

We couple an Acoustic Lens to our mid-range speaker too. It has an *acoustic impedance.* And so does the horn that our high-compliance woofer feeds through.

Between them, you get two *acoustic cut-offs* that match our crossover network. A very uncommon refinement.

It means you *never* get peaks or dips as our three-way Grenadier systems switch from woofer to mid-range to tweeter. *Never* get a *forte* or a *pianissimo* where the score reads *piano,* or a random *sforzandi* accent on a level-volume chord change.

And you never hear music that's muddied up by hums, buzzes, rattles and booms, because our housing stays rigid and firm where boxes shudder and vibrate.

Wide-angle dispersion, smooth, level response and pure, unadulterated music.

Can you think of three *better* reasons to audition a pair of Grenadiers?

GRENADIER SPEAKER SYSTEMS

The Royal Grenadier • $299.95 The Grenadier 7000 • $209.95
The Grenadier 5000 • $179.95

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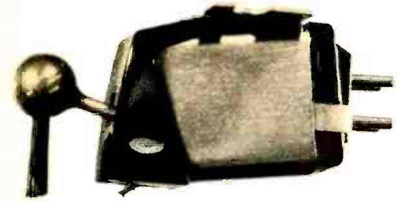
Stanton Model 681EE Stereo Cartridge



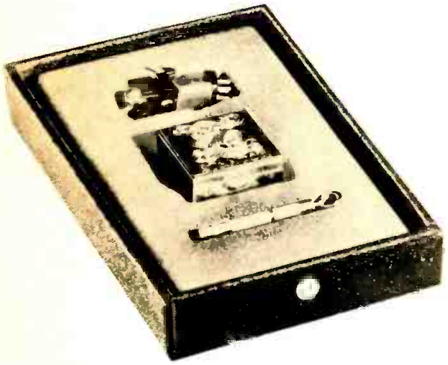


Fig. 1—Stanton "Longhair" cartridge with case, styli container and screwdriver shown below.

MANUFACTURER'S SPECIFICATIONS— Frequency Response: 10 Hz to 10 kHz ±1/2 dB; above 10 kHz, individually calibrated: (test sample) 10 kHz to 15 kHz ±1 1/2 dB; 15 kHz to 20 kHz ±2 dB. Channel Separation (1 kHz): 35 dB. Load Resistance: 47,000 ohms. Stylus Tip: Diamond, .0002 x .0009 elliptical. Tracking Force: 3/4 to 1 1/2 grams. Brush Weight (self-supporting): 1 gram. Cartridge Weight: 5.5 grams. Price: $60.00.

The Stanton Model 681 stereo phono cartridge is the company's top-of-the-line unit. Each cartridge is packaged with individually calibrated performance data, a knurled screwdriver, and an attractive metal "pill box" (for spare styli).

A noteworthy feature of this cartridge is its "longhair" brush, which keeps lint and dust out of the record groove and, naturally, away from the stylus during play.

The stem of the brush is hinged on an off-center pivot, so that it always stays a few record grooves ahead of the stylus point. The bristles also act as an anti-skating device to some degree. In tone arms already employing anti-skating compensation, the arm's compensation must be reduced by about ½ gram to take into account the action of the brush. The bristles, incidentally, never exert a force greater than 1 gram.

The 681's low-mass stylus assembly is probably responsible for the cartridge's superb tracking performance at such low forces as 1 gram. (Perhaps there's an assist from the brush.) In an SME tonearm set for 1¼ grams (plus 1 gram to compensate for the brush), we found that the Stanton 681EE tracked some previously "unplayable" records (for example, a *Project 3* recording, produced by Enoch Light, with Robert Fine as Chief Engineer).

The Stanton 681EE is certainly a smooth one, too. Its frequency response, as plotted in Fig. 2, shows a wide-range response that is free of peaks. Even the usual high-frequency resonant peak is well damped. Response measured within ±2 dB through the 20 Hz to 20 kHz range, which falls within the specifications supplied by the manufacturer as calibration data. Sensitivity was measured as 4.4 mV left and 4.0 mV right, referred to 3.54 cm/sec rms at 1 kHz, which also conforms to the manufacturer's data. Average separation at 1 kHz measured 30 dB, using a CBS STR-100 test record. This is the *best* channel separation figure at this frequency that we've measured over the years. Though our measurement is below the specified 35 dB separation figure (which is 5 decibels more than any other manufacturer claims), one should recognize that only a tiny deviation in pickup alignment (which cannot be avoided without using precise laboratory tools) can reduce the reading by a few decibels. Furthermore, the test record's literature states "to over 30 dB" separation, which may well be only 30.5 dB for all we know.

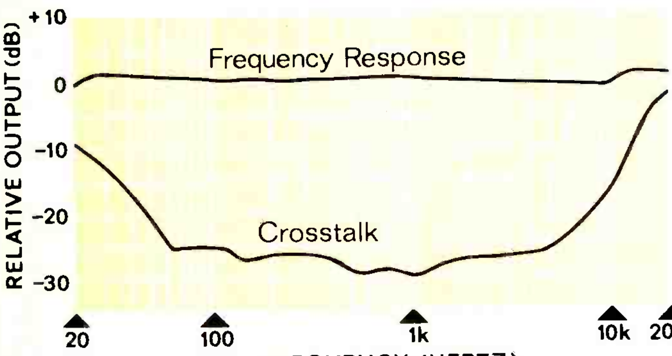


Fig. 2 — Frequency response and channel separation (two channels averaged since they were so close).

The lack of peaks in the 681EE's response is evident in the square-wave photos of Fig. 3. The slight rounding at the top is due to a minor rise in low bass response, while the small wiggle is caused by the very gentle dip at 10 kHz.

With regard to hum-bucking capability, the cartridge's signal-to-noise ratio measured −64 dB through a wide-band RIAA preamplifier. This is an excellent figure, illustrating why the 681EE is not at all susceptible to hum pickup.

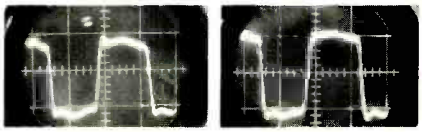


Fig. 3—Square-wave photos; left channel, 5 cm/sec and 3.54 cm/sec velocity.

Performance

The Stanton 681EE was a pleasure to listen through. It brought new life to some old favorites, performed marvelously well with new, bright stereo releases, eliminated the "fuzz" that accompanied some of the heavily cut records.

High frequencies, as produced by brush work on cymbals, for example, exhibited a realistic airiness with the Stanton 681EE. Lows and middles were equally natural, without any noticeable favoring of particular frequencies. The stereo effect was pronounced; this was especially observable with an old *Quarante Cinq* 45 rpm, 12-in. stereo record on bullfight music.

There are many things that measurements cannot reveal, of course. For example, one has to *listen* to determine the degree of coloration produced by a transducer. The Stanton 681EE is a neutral-sounding stereo cartridge, the type of sound we favor, frankly. Discs sound absolutely great when the source material is good and the stereo playback equipment is excellent. Should either be deficient, however, the cartridge will not mask it.

The 681EE stands among the top few cartridges on the market. It would make a fine mate for any of the better automatic turntables, as well as for manual turntables. For turntables that cannot accommodate the lighter tracking force required of elliptical-stylus-equipped cartridges, one can choose the Stanton 681A conical-stylus stereo cartridge. The latter, priced at $55.00 (compared to the 681EE's $60.00), requires 1 to 3 grams of tracking force. This compares to ¾ to 1½ grams for the 681EE.

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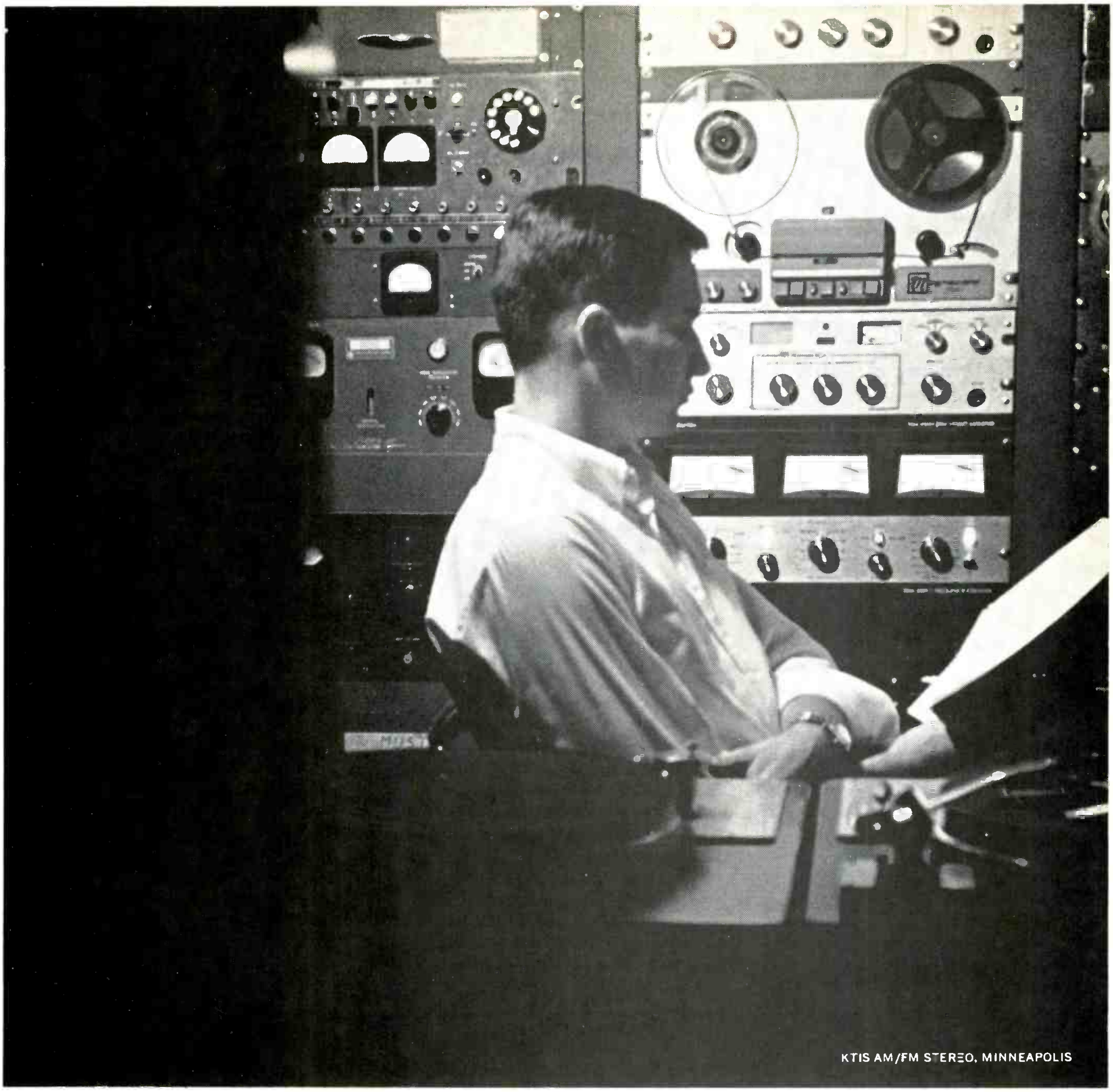

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Magnecord tape recorders run hour after hour, every day, under the toughest broadcast conditions. The die-cast mainplate assures permanent mechanical alignment. Timing accuracy is constant with the hysteresis synchronous capstan tape drive. Payout and take up reels have their own heavy duty split-capacitor motors. In short, solid state Magnecords are built to take it, day after day. You can depend on it.

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MODEL 1024. Commercial or personal stereo unit. Inputs Per Channel: Hi-Z microphone, mixing bridge, auxiliary bridge. Full mixing facilities. 1K ohm emitter follower output. Choice of speeds and head configurations.

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JBL Model SE400S Energizer (Stereo Power Amplifier)

MANUFACTURER'S SPECIFICATIONS— Power Output: 40 watts rms/channel, 10 to 30,000 Hz @ 8 ohms. Frequency Response: With equalized boards set in "flat" position: ±0.25 dB 20 to 20,000 Hz; ±1.5 dB 3 to 175,000 Hz. Harmonic Distortion: Under 0.15%, 20 to 20,000 Hz at 80 watts and less. Intermodulation Distortion: Under 0.15% at 80 watts or less. Sensitivity: 0.8 to 3.0 volts input for rated output, depending on energizer boards used. Hum and Noise: 90-dB below rated output. Dimensions: 4 5/8" H x 15 1/4" W x 7 3/4" D. Weight: 22 pounds. Price: SE400S, $300; SE408S (without cabinet), $276.

The JBL Model SE400S stereo power amplifier features distinctive styling and impeccable workmanship. This Energizer (the name applied by JBL) comes in two styles: Model SE400S is furnished with an olive cabinet for freestanding use. When the Energizer is to be used with a JBL loudspeaker system, however, it can be mounted directly into the back panel of the speaker enclosure. For this application, the unit is supplied, less cabinet, as Model SE408S.

The all-silicon-transistor power amplifier is relatively compact for an 80-watt rms unit, transistorized or not. The front panel does not have any connection points or controls, simply a small lamp indicator behind JBL's logo to show when the unit is on. Power connections, a.c. line fuse, input level controls, and input/output connectors are located in a recess on the rear panel.

The loudspeaker terminals merit special mention. They are spring-loaded, color-coded binding posts that are "blind" on one side so that the speaker wire cannot go all the way through the post and possibly short to the other terminals or chassis. On the inside, the power transformer occupies one-third of the chassis space, and probably accounts for half the amplifier's weight. Just a quick glance at the inner chassis works is enough to convince one that quality of parts and labor is extraordinarily good. For instance, all resistors are of 5% tolerance.

One of the Energizer's innovations is its plug-in equalizer boards. (We're seeing more and more matching of speakers to amplifiers, or vice versa, nowadays.) The equalizer boards have provisions for inserting either end into the contact strip. In one position the equalization is defeated, and the Energizer operates as a standard flat power amplifier. The JBL company supplies equalizer boards for all standard JBL speakers, as well as for some popular systems of other manufacturers. Thus, if the loudspeaker or the acoustic environment requires raising or lowering of a certain frequency range, the equalizer boards can be tailored to accomplish this; the damping factor can be changed via the same technique, although this feature is less likely to be used.

The circuitry of the JBL Energizer is every bit as elegant as its outside design. The output section consists of a three-stage, cascaded emitter follower. (Because of the NPN-PNP configuration there is a resemblance to a "bridged T circuit," hence, JBL's appellation: T-circuit.) There is a single feedback loop from the output back to the base of the first transistor, thus enclosing the whole d.c.-coupled amplifier. A capacitor at the input is the only coupling capacitor in the unit. The output is taken off through a coil, in shunt with a resistor, which effectively prevents high-frequency parasitic oscillations if used with an electrostatic speaker load.

Since the output transistors have a d.c. dissipation rating of 150 watts each, no esoteric protective devices are needed and none is used, except for a pair of thermal circuit breakers located

Fig. 2—Frequency response (at 1 watt) and power bandwidth (60 watts, at 8 ohms, both channels driven) exemplifies JBL's wide-band design philosophy.

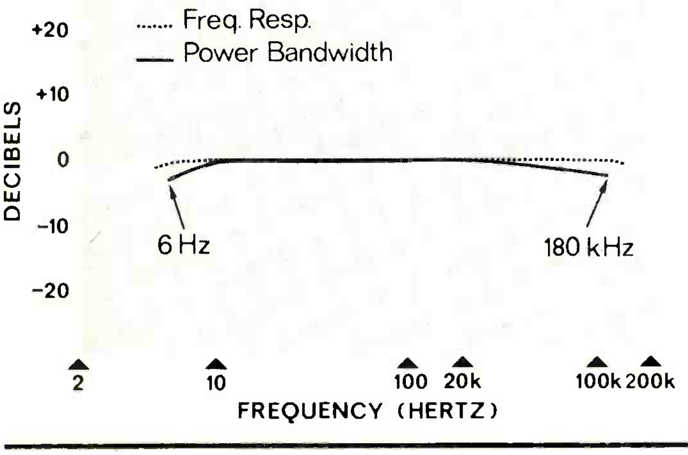
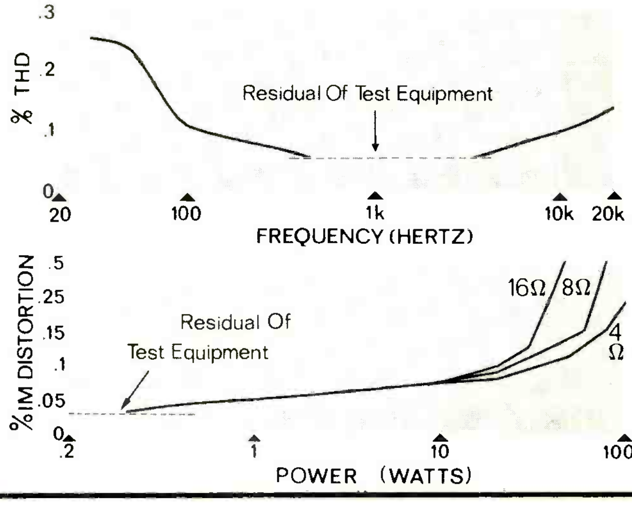


Fig. 3—Total harmonic distortion was under 0.25% at 60 watts, into 8 ohms, both channels driven. Distortion measurements vs. power with a 1-kHz signal would have left us with a chart that was below the residual of our test instruments, 0.06%. IM distortion is extremely low, too, as can be seen here.



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on the heat sinks. These will open if the amplifier has overheated due to poor ventilation or a short circuit at the loudspeaker terminals.

Performance

As a look at the performance curves (Figs. 2 and 3) shows, the specifications are extremely conservative. James B. Lansing rates the SE400S at 40 watts per channel into 8 ohms. We measured *60 watts* at below 0.25% distortion, with both channels driven simultaneously. At 17 Hz and 35,000 Hz, total harmonic distortion reached 0.5% at 60 watts; at 1000 Hz we read the residual of our instruments, 0.06%. Power response, as monitored on an oscilloscope for waveform purity, extended from 6 Hz to 180,000 Hz, thus being almost as wide as the 1-watt frequency run, which went on to 270,000 Hz before being down 3 dB. We had to go to the extremes of 20 Hz and 20 kHz in the square-wave test to find any worthwhile discrepancy between the input and output waveforms. See Fig. 4.

It turns out that JBL could safely rate the Energizer at 50 watts per channel into 8 ohms and still be on the conservative side. Power output was

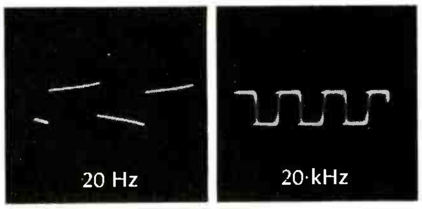

Fig. 4—Square-wave response at 20 Hz and 20 kHz.

unaffected by normal line-voltage variations. Whether one or both channels were driven, the output power available remained the same, which indicates a generous power supply. The signal-to-noise ratio relative to 60 watts was 94 dB—again, better than the specifications.

When it came to the intermodulation distortion measurements, the amplifier was in a class by itself. With a 4-ohm load and both channels driven, it delivered 95 watts per channel at 0.25% IM distortion. More important, at lower levels the distortion kept falling until we were eventually limited by the residual of our IM analyzer (which cost almost twice as much as the amplifier), .035%.

The internal impedance, as measured at the output terminals, is 0.3 ohms, which gives a damping factor of 26 to 27 when connected to an 8-ohm speaker. Overload recovery time is supposed to be less than 1/10 of a cycle from 20 to 20 kHz. Unfortunately, our test setup during the period we had the amplifier was not up to making this measurement. Therefore, we had to rely on our ears to see whether any clipping would be audible. Suffice to say that driving a pair of low-efficiency, 3-way acoustic-suspension speaker systems in a well-damped room, our ears gave up before the amplifier showed signs of distress. Using a full-range electrostatic, the amplifier worked equally well, displaying great stability.

Music reproduced through this amplifier was as clean as we have ever heard through any amplifier. The JBL SE400S is intended for home use, we know, but it would do equally well in studio or laboratory applications, judging by its fine performance and sturdy construction. The price of $300 is not inexpensive for a stereo power amplifier, but this one is well worth it. We have only one question: What can JBL do for an encore?

Check No. 56 on Reader Service Card

Should you be a nitpicker...

Should you be a nitpicker when it comes to selecting a stereo deck? Only if you want to get yourself a deck you'll be happy with for years to come.

Because every manufacturer *claims* to have the "guts" to make the best sound. But, if you had the opportunity to "tear apart" most of the tape recorders on the market, you'd find a lot of surprises inside.

Like flimsy looking little felt pressure pads to hold the tape against the heads which actually cause the heads to wear out six to eight times faster than Ampex heads.

Like stamped sheet metal and lots of other not-so-solid stuff that gets by but who knows how long? And all kinds of tiny springs and gadgets designed to do one thing or another. (If you didn't know better, you'd swear you were looking at the inside of a toy.)

Like heads that are only adequate. Heads that might work fine at first, but wear out sooner and diminish the quality of sound reproduction as they wear.

There are lots of other things, but that's basically what *not to get* in a deck.

Okay, now for a short course in what *to get*.

Exclusive Ampex dual capstan drive. No head-wearing pressure pads. Perfect tape tension control, recording or playing back.

Exclusive Ampex rigid block head suspension. Most accurate head and tape guidance system ever devised. Solid.

Exclusive Ampex deep gap heads. Cost about $40 each. Far superior to any other heads on the market. Last as much as 10 times longer. There's simply no comparison.

So much for the "general" advantages of Ampex decks. Ready to nitpick about *specific* features on *specific* machines? Go ahead. Pick.

Pick the Ampex 755 for example. (This is the one for "professional" nitpickers.) Sound-on-sound, sound-with-sound, echo, pause control, tape monitor. Three separate Ampex deep gap heads.

Or, pick the 1455. For lazier nitpickers, because it has automatic two-second threading and automatic reverse. Plus sound-with-sound, pause control and tape monitor. Four separate deep gap heads.

One more thing you should get on your next deck, whichever one you choose: the exclusive Ampex nameplate on the unit. Just big enough to let everybody know you've got the best. (Who says a nitpicker can't be a name-dropper too?)

So, pick, pick, pick. And you'll pick Ampex. Most straight-thinking nitpickers do, you know.

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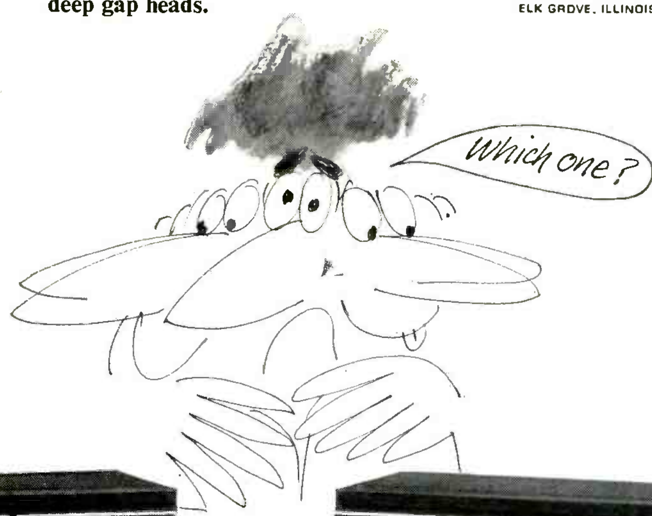

Model 755

Model 1455

A deck for nitpickers.

And a deck for lazy nitpickers.

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BUILD YOUR OWN Presence Control

AL FANNING

IF YOU LIKE recorded vocal music and own many records featuring vocal selections, this project is for you! Ever wonder why, in some of your recordings, the vocalist seems masked or buried by the accompanying orchestra, while in still others, the vocalist is "right up there" at stage front-and-center? The answer lies in the recording studio control room where, in all probability, the recording engineer has electronically created the "stage presence" you find so lifelike. The device he uses to create this effect goes by various technical names which we prefer to call a "presence control."

Although the human voice contains low, middle, and high frequencies much like a musical instrument, it has been found that the greatest vocal energy lies in the range between about 400 and 1500 Hz. By emphasizing these frequencies somewhat during playback, it is possible to bring the vocal portion of any program into pleasing prominence. Obviously, a conventional tone control on your amplifier won't do. If you were to boost your treble control, all frequencies above 1000 Hz would be emphasized, resulting in screechy highs. Boosting your bass control will add boom. Boosting *both* tone controls has an effect just opposite to the desired one. Highs and lows are both accentuated while the vocalist retreats backstage. To accomplish improved presence, we need to boost only the mid-frequencies while we leave all the others alone.

It doesn't take much electrical emphasis of these mid-frequencies to accomplish a startling improvement in "presence." In practical terms, if we boost the "voice frequencies" by only 6 dB, the vocalist will sound half as far away from you as the orchestral accompaniment.

Professional filters used for presence controls generally use expensive coils in tuned circuits as part of the system. Because these coils are expensive and are not readily obtainable, the presence control to be described uses only resistive and capacitive elements. A schematic of the control for one channel is shown in Fig. 1, and the wiring diagram is shown in Fig. 2. A dual potentiometer, a few resistors, and two capacitors are all that you will need if you want to wire the control into an existing amplifier or receiver. For a stereo system, of course, you will need two separate controls, although the parts list shows how you may order the components to make the control work on both channels simultaneously. Either way will work, but the four-gang control is easier to use. If you cannot wire the unit into your present amplifier or receiver, you can mount it readily in a 2¼" x 2¼" x 4" Minibox, and use shielded cables to connect it between preamp and power amplifier as shown in Fig. 3.

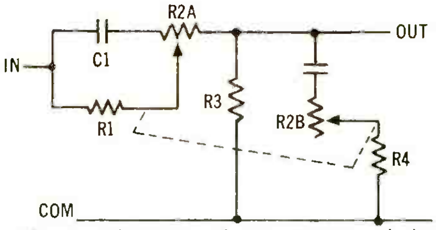

Fig. 1. Schematic of one section of the simple presence control. Two sections are required for stereo.

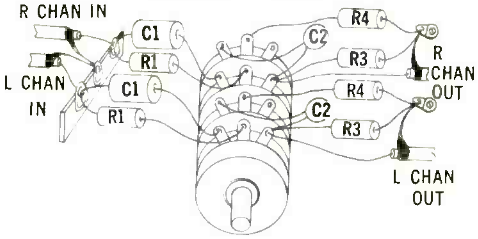

Fig. 2. Wiring diagram for the four-gang potentiometer used in making the stereo model of the presence control.

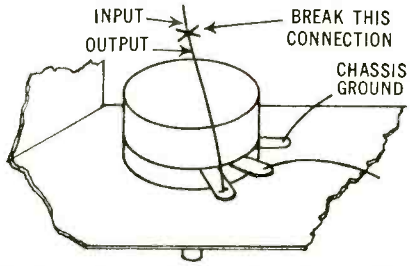

Fig. 3. Method of connecting one section of the presence control to a monophonic volume control. For a stereo volume control, duplicate the connections to both sections of the potentiometer.

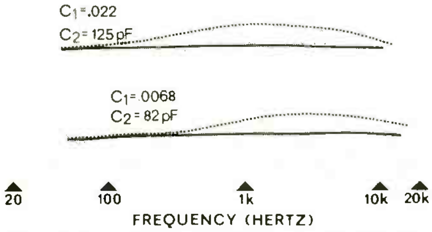

Fig. 4. Frequency response curves for the presence control with the components specified, together with the curve when the components C_1 and C_2 are changed to .0068 μF and 82 pF respectively.

PARTS LIST

C_1—.022 μF capacitor, Sprague 192P22392

C_2—120 pF capacitor, Sprague 10TCC-T12

R_1, R_3, R_4—47 k ohms, ½ watt

R_2—100 k-ohm, 4-gang, log-taper pot (one IRC A-104 and 3 IRC MA-104 sections).

Miscellaneous: control knob, control mounting hardware, terminal strips, wire, solder, etc.

(Note: For a stereo system, two each of items C_1, C_2, R_1, R_3, and R_4 are required.)

Wiring the Presence Control

In following the wiring diagram of Fig. 2, be especially careful of the wiring to the terminals of the four-gang potentiometer. As shown, the increase in presence occurs as the control is rotated clockwise.

If you own a preamp-amplifier combination or a complete receiver, the best place to "break into" the circuit is at the high side (input) of your volume control, and in the case of stereo systems, this must be done twice, once for each channel. Most single volume controls are connected as shown in Fig. 3, with stereo controls having two sets connected in the same way. The wire(s) going to the ungrounded end of the control should be disconnected and fed to the input of the presence control. The output of the presence control should then be connected to the control terminals from which the wire(s) were disconnected. Figure 4 shows the frequency response of the control in the flat (counterclockwise) position in a solid line, and in the presence position as a dotted line. The point of maximum "boost" is at 1300 Hz with the parts specified. Changing C_1 and C_2 to .0068 μF and 82 pF respectively will raise the point of maximum boost to about 3200 Hz.

To try out the presence control, play one of your favorite vocal recordings and start with the control at the counterclockwise position, and listen carefully to what happens as you rotate the control clockwise. While a boost of only about 5 dB may not seem like much, you will be surprised at the effect. Æ

PAT-4 PREAMPLIFIER
89.95 KIT, 129.95 ASSEMBLED

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LASTING QUALITY

In the evolution of high fidelity, there have been some "revolutions"—the stereo record, FM multiplex, and transistorization, to give some examples. Each of those changes left its trail of obsolete equipment, frequently replaced with much higher priced models. Through these periods of change, Dynaco has maintained a level of quality so high that our equipment is always current, never obsolete, and always adaptable to the newest useful innovations.

Dynaco's underlying philosophy is to deliver exceptional performance from designs so carefully and progressively engineered that they defy obsolescence. We add new products only when we feel that they can make a contribution of value to music reproduction. In each Dynaco high fidelity component the total value of the separate parts is greater than what you pay for the finished product, and you can save even more by buying the kit.

Dynaco's separate components give you the ultimate in flexibility and ease of installation. They can be interchanged with full compatibility, not only with Dynaco units, but with any other similar designs which are generally accepted as being of the finest quality. No industry innovation can make your **system** obsolete, and future changes, such as an increase in amplifier power, can be easily and economically accomplished.

The quality of performance obtained with the FM-3 tuner, PAT-4 preamplifier, and the Stereo 120 power amplifier cannot be matched in any single package regardless of promotional claims. Other Dynaco units which can interchange with this system will also give similar results at lower power, or with a bit less control flexibility at still lower cost, depending on the units chosen.

Whether you compare Dynaco with others by listening or by laboratory test, you will find that Dynaco gives sound closest to the original—with lucid clarity, without murkiness, noise or distortion. Every unit—whether purchased as a kit or factory assembled, is assured of delivering the same specified quality, for our reputation has grown through directing our design efforts towards perfection rather than to the planned obsolescence of yearly model "face-lifts."

You may find that your dealer does not have some Dynaco equipment in stock, however, for the demand greatly exceeds our ability to produce for a rapidly growing audience. Quality is our first consideration, so we must ask your patience. We believe you will find it is worth the wait.

STEREO 120 AMPLIFIER
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Electronic Organs

NORMAN H. CROWHURST

PART 4 of a series. How formant circuitry is used to shape the tone signals to simulate organ tones.

PREVIOUS INSTALLMENTS introduced the problem of getting "body," or quantity of sound in an organ. To develop the problem, we will now get down to detail. The least expensive way to put an organ together is to use a set of master oscillators and frequency dividers. These outputs are then keyed, and the output from the keyboards, either before or after amplification, is fed through formants controlled by the tabs.

Many modern organs improve the effect of some of their individual voices by treating sections of notes separately, usually about an octave at a time. This is particularly necessary for formants that change the harmonic content appreciably. For example, to get a flute tone, the upper harmonics are "smoothed off."

The upper harmonics of a note in the lower octaves would be attenuated by a whole-range formant, but the higher fundamentals would also be attenuated by a like amount. Therefore the upper notes would be weak. If the whole output is put through a formant that produces, say, a 6 dB/octave slope, the lower notes are going to be very much louder than the higher ones. If this is corrected by an equalizer, the harmonic structure is put back to its original, and the flute tone is lost.

So each octave of output is separated and passed through a separate formant (Fig. 4-1). If you listen carefully when playing the scale one note at a time, with only one tab in action, you may hear where the change occurs due to transferring to another formant (in the form of a change in level). In this case, as you go up the scale the notes will get imperceptibly quieter and then, at one particular note, they'll get louder again.

You may find this more noticeable if you play notes in octave pairs, and go up the scale in this way. But when the organ is played normally, several notes at a time, or with more than one tab working, or both, this difference is not noticeable.

Another method of obtaining overtone-reduced sound to approximate flute sound uses L-C filtering with half-octave sections (Fig. 4-2). By using a pair of L's and a pair of C's in staggered combinations, each successive cut-off, from section to section, advances half an octave. Outputs from six notes are injected at each junction of the filter. This results in a smoother tone—greater reduction of harmonics, with less attenuation of fundamental. The change of level as notes in consecutive sections are played is negligible.

Yet another approach to flute voicing selects fundamental to emphasize, rather than eliminating harmonics. Thus the full range of harmonics is there, but fundamental is slightly emphasized over them. This is done over the mid-range more particularly, using filters, for example, at 425, 600, 850

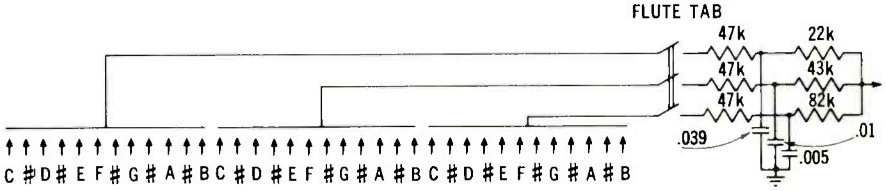


Fig. 4-1—Dividing the keyboard for application of formants helps maintain uniform level from the whole organ.

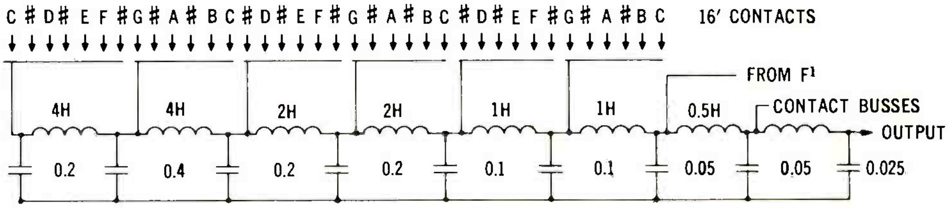
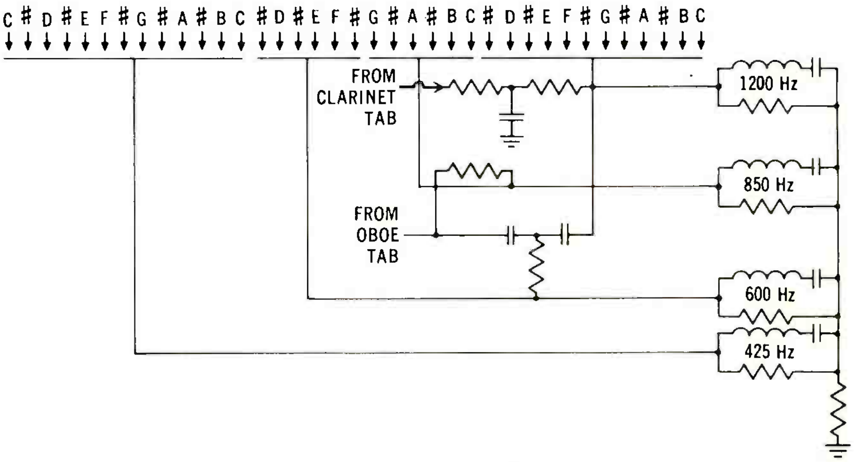


Fig. 4-2 (above)—A more sophisticated way of achieving flute effect than the simple circuit of Fig. 4-1.

Fig. 4-3 (below)—Yet another way of achieving the flute formant. Here other tabs use part of the same networks, but for the whole keyboard in that case.



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For example, here's what you get in the new AD-27: Quality electronics from the Heathkit AR-14 Stereo FM Receiver, modified only mechanically to fit. Audio Magazine said about the AR-14: "... undoubtedly one of the best values we have encountered to date." Plus the precision-crafted British BSR McDonald 500 Automatic Turntable, complete with famous Shure diamond stylus cartridge. And a beautiful walnut cabinet with sliding tambour door to house them. Here's the AD-27 in detail: Amplifier portion features: • 15 watts per channel music power — ample to drive almost any reasonably efficient speaker system • Solid-state circuitry • 12-60,000 Hz ±1 dB frequency response • 45 dB channel separation • Harmonic & IM distortion less than 1% • 4 to 16 ohm output impedance • Tape output • Front panel headphone jack • Speaker defeat switch for private listening. FM Stereo section features: • 4 stage IF gives 5 uV sensitivity for real station pulling power • Hum & Noise —45 dB • Less than 1% distortion • Smooth inertial flywheel tuning • Adjustable phase control for best channel separation • Stereo indicator light • 20 dB channel separation • Filtered outputs for "beat-free" stereo taping. High quality BSR McDonald 500 Automatic Turntable with these features: • Low mass tubular aluminum tone arm • Adjustable anti-skate control • Micrometer stylus pressure adjustment • Cueing/Pause control • Diamond Shure cartridge with 20 Hz to 20 kHz response • Plays all 4 speeds — automatic, semi-automatic or manual • 4 pole induction motor • Low wow and flutter.

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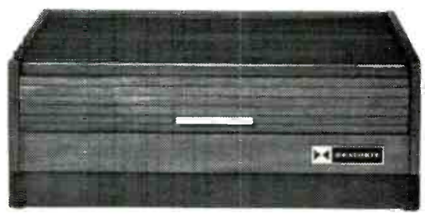

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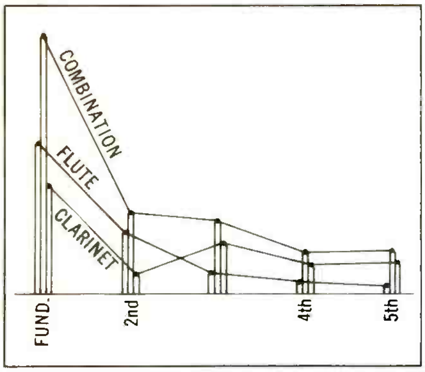
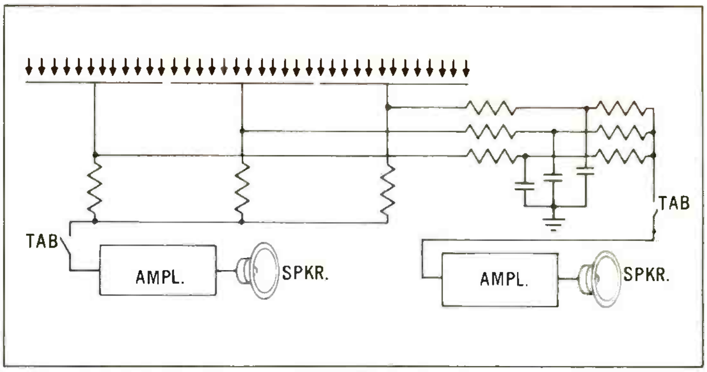


Fig. 4-4 (left)—What happens when tabs using formants only are combined (see text).

Fig. 4-5 (right)—One way to achieve a slight improvement when the organ has more than one amplifier and speaker on its output.



and 1200 Hz (Fig. 4-3) for half-octave blocks of notes in that range. The bottom and top sections extend further than half an octave. Some organs that use this filter method use the same filters for other purposes to effect an economy in components. For example, oboe and clarinet, with different additional equalization, may use one or two of the filters. Here the clarinet uses one and the oboe two of them, each with additional equalization.

The methods of achieving formants for different tones are virtually infinite. An organ maker tends to stay with a particular method of achieving his favorite voices, until he discovers what seems a better way.

Now let's look at what happens when a single note is pressed, and different tabs are tried. Fig. 4-4 shows the situation. One tab selects, say, a formant that gives a good imitation of clarinet sound. The frequency response of the network through which the harmonic-rich wave is passed changes the relative harmonic amplitudes so that the output sounds like a clarinet. Now cancel that tab and use one that selects a formant that gives a good imitation of a flute. Changing from one to the other, you are impressed by the realism with which the instrument imitates these two different sounds. Now you press both tabs together, so both formants are in use together. What do you get? In Fig. 4-4, the vertical columns at the left indicate the relative amplitudes of fundamental and harmonics for the flute sound. Those at the right indicate the relative amplitudes for the clarinet sound. The central columns add the two, to show the relative amplitudes when both tabs are used together. As shown in Fig. 4-4, it would be a little louder, but this may not necessarily be true because the paralleling of the formant circuits inputs and outputs will cause them to shunt their impedances mutually so that the overall transmission is not completely additive. However, the combined response is some kind of compromise between the two formants, obtained by summing the transmission through each.

Whatever this sounds like, it is still only one note. It is something *between* a clarinet and flute, rather than sounding like the two put together, which is what we expect to hear if we don't think about what's inside the organ. How can we change this? This is one problem confronting organ designers.

If the outputs of the individual formants are fed to different amplifiers and speakers (Fig. 4-5), spatial separation would be provided between the two sounds. This is a little better. The

(Continued on page 71)

Fig. 4-6—A way to provide tremolo on some stops and not on others, at the same time. This improves the body to make two stops sound like separate sounds.

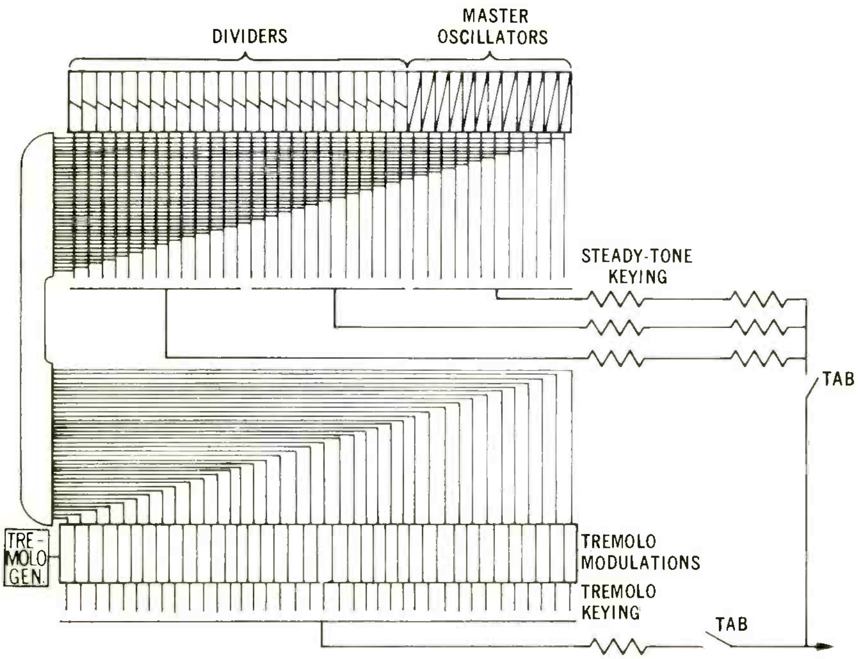
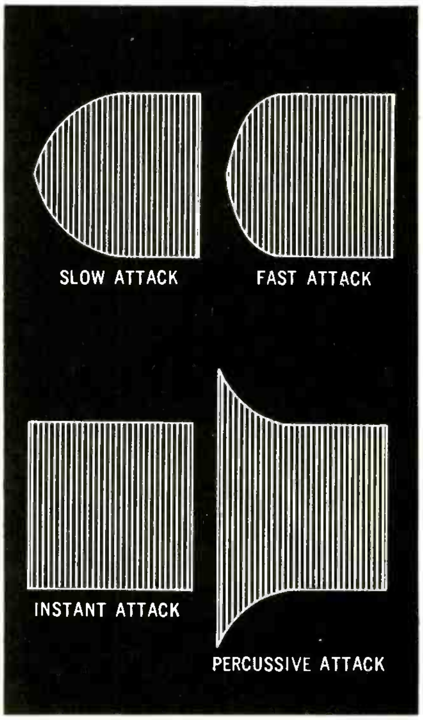


Fig. 4-7—Different attacks possible, when keying is more than a simple "on-off."



IF THE CROWN CX822 IS JUST ANOTHER TAPE RECORDER

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CX822

The Crown CX822 has been opening eyes in testing labs all over America. In early 1968, Audio magazine put it to the test, and published its findings in the April Equipment Report. Following are a few excerpts from that report:

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"We found the tape motion command system to be as foolproof as Crown says it is, and could not beat the computer by design or by accident."

"The CX822 is a serviceman's dream, considering the unit's inherent complexity."

"Playing back first-generation transfers from original masters, the sound produced through the Crown CX822 was peerless. When recording and playing back from records and FM broadcasts, there was absolutely no aural difference between the original and the copy at 15 ips. The same held true at 7½ ips, though theory says, there should have been."

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"The new Crown CX822 is capable of providing the most faithful reproduction of sound through the magnetic recording medium that we have observed to date. And it does it in as foolproof and as easy a way as we've seen."

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The 2nd Coming of ERIK SATIE

EDWARD TATNALL CANBY

That embattled French musical revolutionary and wit of yesterday, who would be 102 today if he had not died in 1926, the goat-bearded, rimless-spectacled Erik Satie (see drawings by Picasso, Cocteau) has made a sort of belated second debut these last months, before the great modern public that listens to its music via LP records. About time—for very few of us other than musical professionals had ever come to know this man in actual musical sound, though we've always heard about him in second-hand descriptions.

Angel, updating a Volume 1 of Satie piano works into brand new stereo, has added two more discs, Vols. 2 and 3, six full sides of Satie played by the superbly understanding Aldo Ciccolino, a most revealing musical "one man show" with never a dull moment on any of the sides. Especially if you note the amusingly outrageous titles that gave these little pieces some of their claim to notoriety. Along with these, Angel offers a complementary orchestral disc from the home town, Paris, including the two once-celebrated ballet scores, *Parade* (1917) and *Relâche* (1924), plus Debussy's fond orchestrations of two of the little piano *Gymnopédies* of 1888, tiny morsels that first suggested the Impressionist language that Debussy was to make his own.

Finally, there is Vanguard's sumptuous new album, "Homage to Erik Satie" on the middle-price Cardinal label, an all-orchestral gate-fold affair with two records that include not only the same two ballets, for an interesting contrast in interpretation, but a third ballet score, "The Adventures of Mercury," also from 1924, and a whole group of short orchestrations of Satie piano works by various of his admiring colleagues and followers. Darius Milhaud, one of the few still-living composers whose musical careers were directly shaped by Satie's influence, was enthusiastic about his last album. An article by him appears as part of its liner notes.

There are two sorts of aesthetic radicals of the *avant garde* in any time—those who are fanatically determined, and those who are irrepressibly humorous, in the barbed or practical-joke fashion. The most devastating of this last sort are the deadpan artists, who never crack a smile but manage to pull the establishment's leg even harder for it—we don't quite know, as the old phrase goes, whether to laugh to to cry. John Cage is our current master of this technique, as in his well known piano pieces that consists of absolute silence for a given number of seconds. Audiences are reduced to total confusion by that bit of concert mayhem; but their musical lives may never be the same again afterwards.

Satie, if reports are true, was a gentler thorn in the side of the musical establishment, though by a very small degree. *His* establishment was not quite ours; his battles are now largely won, and so his music often sounds corny and exaggerated in its iconoclasm, as of so many years later.

But don't underestimate his prowess as a composer. He was good. For evidence, listen to Aldo Ciccolini's marvelous performances of the many little earlier piano pieces, the ones with the gadfly names, making fun of the prevailing craze for mysteriously wordy titles—"The Sunk-in Cathedral," "Girl with the Golden Hair," "Reflections in the Water," to name a few by Debussy. Satie's are astringent. "Three Pieces in the Form of a Pear," "Dried Embryos," "Picturesque Infantilisms," "Menus for a Childish Purpose," "Flabby Waltzes" and dozens of others. In this middle period, notably in the year 1913, Satie turned out a polished piano product easily in a class with Debussy himself, and indeed sounding remarkably like the late Debussy, though that master is more dissonant and much less humorous.

Debussy, four years older than Satie, did not live to see the more fundamental revolution that now broke out, largely sparked by Satie's indefatigable pin pricks. It was, just as Satie would have wished, a battle to the death against "classical" music of the genteel, overweight German tradition, out of Bach, Beethoven, Brahms, Mahler, even on to Schoenberg. If not to the death, then at least to a point of equality—relegating the older music to its place as merely one way of doing things. In such revolutions, of course, the laws of nature must prevail. To move the immovable takes force. Action and reaction are both equal and opposite. To get at Romantic musical elegance, there was only one possible artistic course—*inelegance,* blatant lowbrowism. Some of the finest minds of the 'teens and twenties worked at *that* peculiar aesthetic problem, with every bit of ingenuity they could muster! The results were devastating, especially in Paris where things Germanic were particularly suspect.

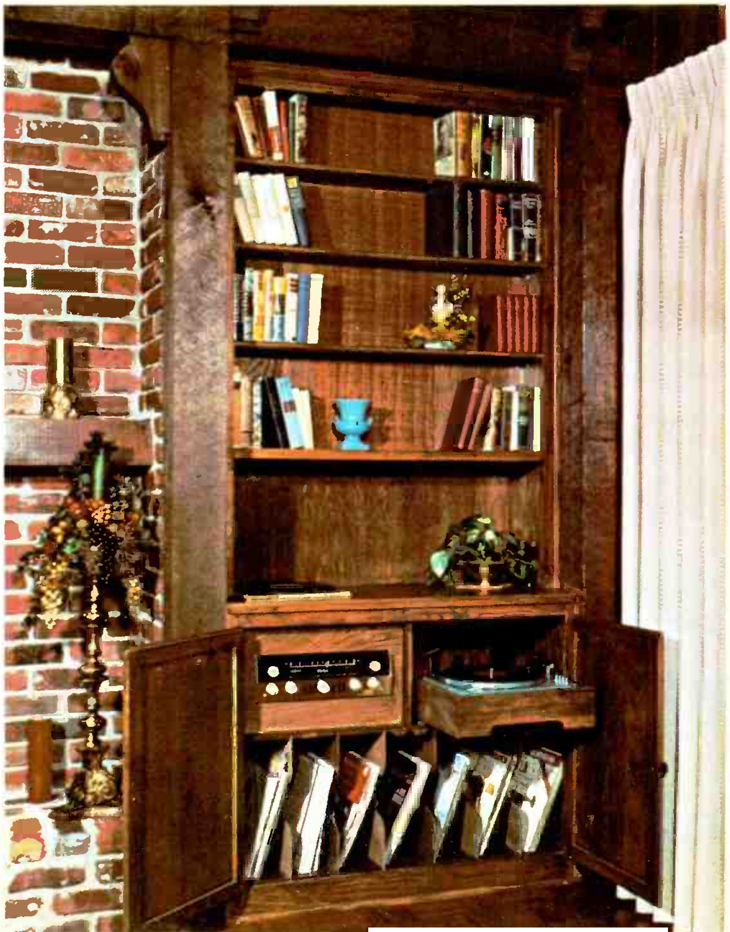
Thus Satie's quarter century of sticking little pins into the establishment suddenly bore wild fruit. Now he had a whole new generation of devoted followers, to lend him scandalous prestige. Middle-aged, he was youth's champion—he worked with Picasso, Jean Cocteau, Massine, Diaghilev and, in music, youngsters such as Poulenc, Milhaud, even the Russians, Stravinsky and Prokofieff (then both adopted Parisians). He dropped his sophisticated Debussy piano style and plunged into the new vulgarity with fiendish glee. The thing was to sound as brassy, as lowbrow, as trivial, as loud and noisy as possible—it was exhilarating fun and *very* unstuffy, as well as definitely un-Germanic.

It worked. We got Gershwin, Virgil Thomson, Copland, Kurt Weill and plenty more now-revered composers. But in that foamy time, does Satie's music sound corny! Unbelievably dated. Even so, you will be delighted with the blats and the blurps, the brassy tunes and the jazzy rhythms, the rattles, shotguns, wood blocks, and general nose thumbing. It's very much alive, corny or no. The old goatbeard must have been irresistible, at least to his friends.

Piano Music of Erik Satie, Vols. 1, 2, 3. Aldo Ciccolini. **Angel S-36482, S-36459, S-36485 stereo $5.79 ea.**

Erik Satie: Parade; Relâche; Gymnopédies (orch. Debussy). Paris Conservatory Orch., Auriacombe. **Angel S-36486 stereo $5.79.**

Homage to Erik Satie. (Parade; Aventures de Mercure; Ralâche; assorted orchestrations by friends of Satie). Utah Symphony Orch., Abravanel. **Vanguard Cardinal VCS 10037/38 stereo $7.**



1.

Sound & Decor Styles

HERE ARE some custom installations by a component dealer, David Beatty, Kansas City, Mo.

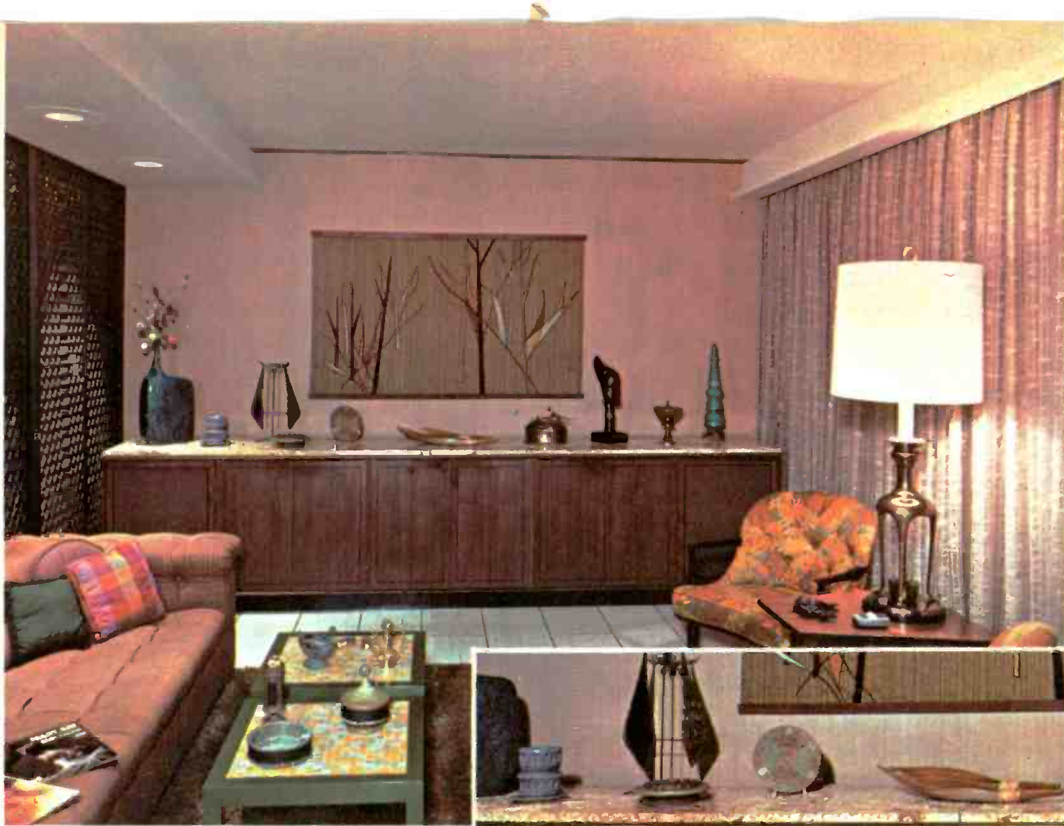
1. Installed for John Ballard, this stereo installation has at least one unique (to our knowledge) feature. It concerns the installation of speakers. (We challenge you to find the speaker locations before reading further.) There are large speaker enclosures behind each bookshelf. Looking at the second photograph, which shows one bookshelf section, speakers are located behind the center of the top four bookshelves. Where, you ask? Observe the slight difference in "walnut" coloration of the middle section. That's the grille cloth you're looking at, actually Italian silk that blends with the walnut so that one is hardly aware of it. The books are placed on the shelf, but are separated in the center, where the speakers are located. A high-frequency horn is situated at about ear level, while a 15-in. speaker is above it.

A McIntosh Model 225 power amplifier is suspended from the basement ceiling, just below the equipment cabinet, where you can see other stereo equipment and LP discs.

The idea of having speakers on either side of a fireplace, built into bookshelves, is very effective from both a decorating and an acoustic standpoint.

2. This room was added on to a house. A custom cabinet was constructed to hold stereo component equipment and a color television receiver. The cabinets have special ducted tube ports to enhance bass frequencies. You can see a Thorens Model 224 automatic turntable in the closeup photo.

3. Here is another stereo component installation in a long cabinet. Speakers are installed in the wall behind the floor-to-ceiling sections at each side of the cabinet.



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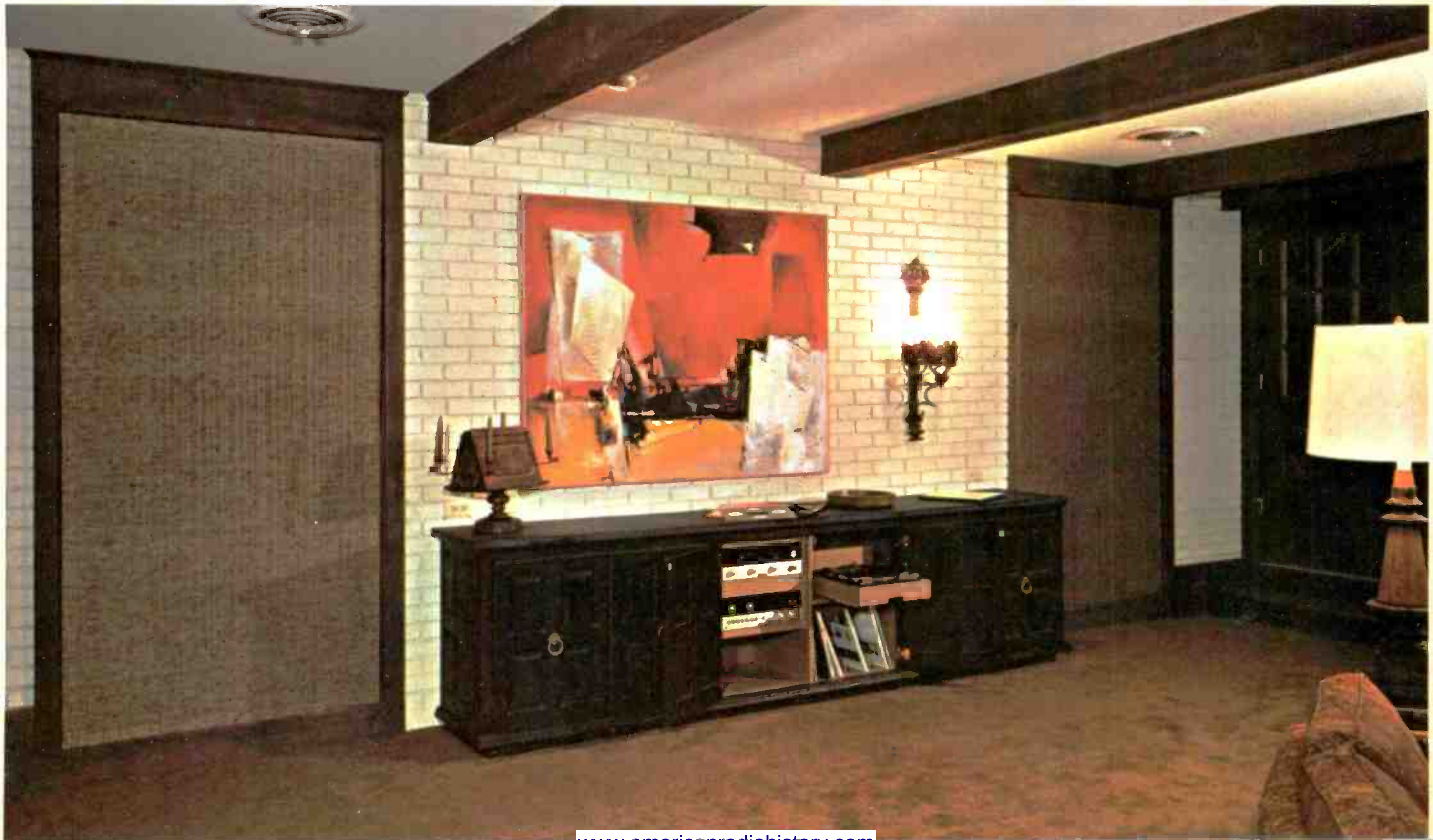


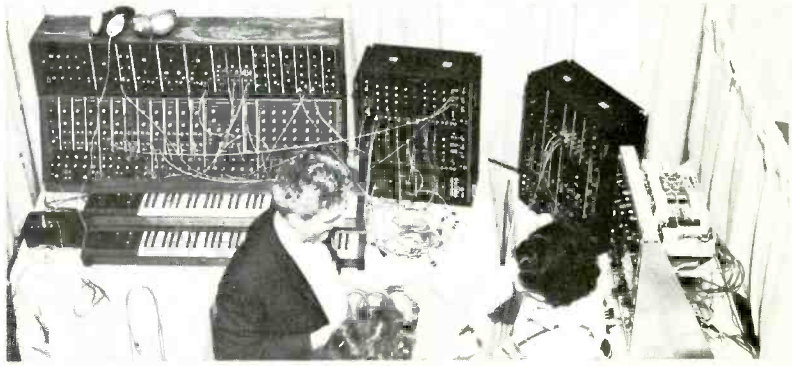
The Audio "Pro's" at Show Time

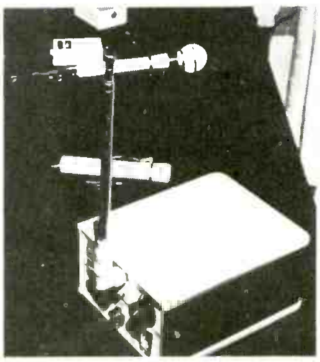
The 35th Audio Engineering Society Convention and Exhibition in New York City celebrated the 20th anniversary of the society's founding with the largest display of professional equipment to date. Shown here are some exhibit scenes captured by Audio's roving photographer.

Scene in the Corinthian Room during the evening seminar, "Audio Engineering and the Environment."

Paul Stone (left) demonstrates Bozak's new CAV-6-2, Perception Mixer.

Robert Moog demonstrates some of his equipment for producing electronic music.





Part of the Sennheiser line showing its wireless microphone systems.

Shure Brothers displayed its complete line of professional and consumer microphones.

Self-contained 8-track Ampex recorder with individual amplifiers.

Badisch Anilin-Soda Fabrik (now you know what BASF means) provided an interesting exhibit which featured a large photo of its German factory.

Ed Straw, New York Crown rep. following the sign's invitation.

1800-foot horizontal tape bin of Gauss Electrophysics with tape speeds up to 240 ips.

Martin Audio "Varispeed III" recorder assembly.

ELECTRONIC ORGANS

(Continued from page 64)

sound can now be imagined as a clarinet and flute playing in unbelievably close unison. And so long as the organ maker is using more than a single power amplifier and associated speaker, to make "more organ" in that sense, feeding individual amplifiers with different parts of the total musical sound does make more sense.

But how can the difference be accentuated, to sound more like two separate instruments played together, or like two sets of organ pipes playing together? Anything that will "loosen" the perfectness of the unison helps. The obvious thing would be a completely separate set of tone generators for each voice. But this would rapidly run up the cost of the organ. Each would need separate tuning, apart from all the extra electronics involved.

You could apply gentle vibrato, or tremolo, to one and not to the other. Tremolo is the easier to do, because it involves changing only amplitude, not frequency. This still involves additional "handling" electronics for every note (Fig. 4-6), so the cost is more than the simple formant combining system. But the extra electronics don't need tuning, as another complete set of generators would.

Maybe it doesn't achieve quite as much, but a way that does a little more for the money involved is to use different "attack" effects at the beginning of each note, as it starts to "speak" (Fig. 4-7). A surprising proportion of the realism in imitating sounds comes from successfully copying the way they start. In this method, the attack must be applied with the keying, and the formant(s) added after.

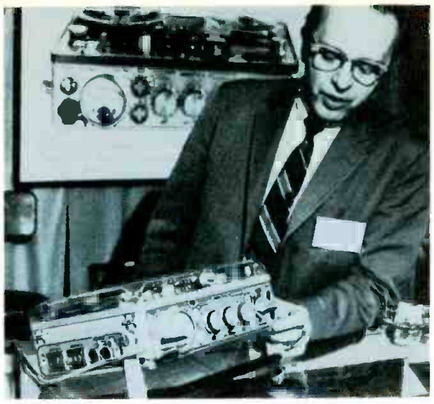
Some organ-makers have even added means to inject tones when the note is first pressed that disappear as the steady note is established. Most wind instruments and the plucked strings show this kind of effect, if you listen carefully.

The first pulse of air, or the release of the string, excites a random, fairly-high-frequency composition from which the pipe or string resonance selects higher overtones of the eventual note to emphasize, while the standing wave pattern that constitutes the true note builds up. This is a special effect that some organ makers have added, and we will go into the electronic details later. But incorporating it relates to the keying. Æ

CONTINUED NEXT MONTH

The Audio Pro's at Show Time (continued)

With multi-track recorders in abundance, Metrotech showed a simple two-track stereo model.

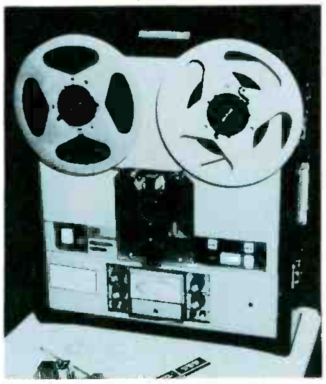


The famous Nagra professional portable shown in a plastic see-through case.

Altec's line of equalizers, filters, and components were shown in front of its studio monitor loudspeaker.

Seen for the first time by most of those in attendance, Norelco's Simultaneous Interpreting System was exhibited, using AKG headphones for the aural demonstrations.

3M film-sync two-track recorder, with one track for audio and one for the sync signal.



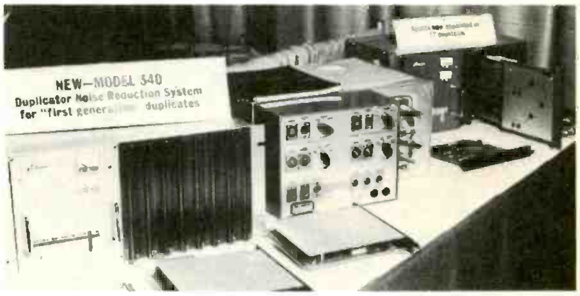
Electrodyne's elaborate 8-channel mixer console offers a wide range of facilities.

Harman-Kardon professional p.a. equipment provides some innovations in reliability.

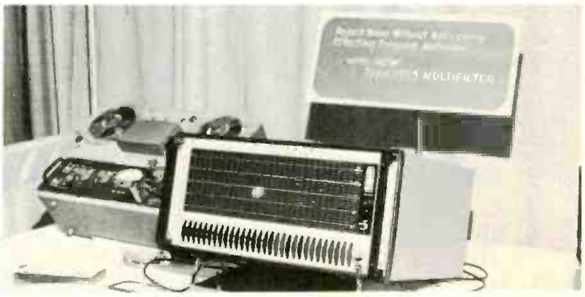
Tandberg was present with their new Model 11 portable, as well as the Huldra multiband receiver.

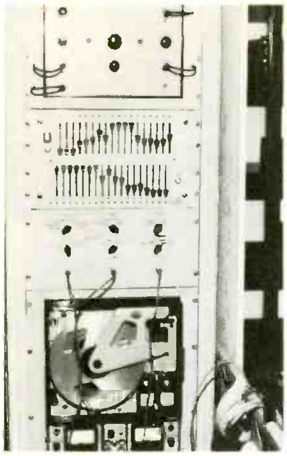
The ideal machine for logging, Tape-Athon's Model 900 Logger.

Dolby introduces his new "Duplicator Audio Noise Reduction System."



General Radio's new Type 1925 Multifilter provides a wide variety of filtering for laboratories and acoustic installations.

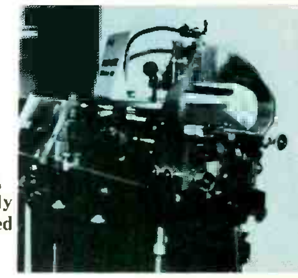




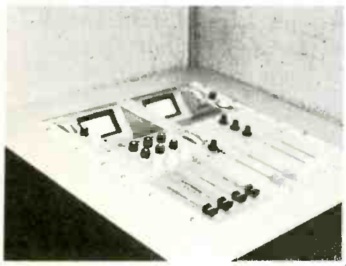
B & K Instruments exhibited Bruel & Kjaer's 36-channel spectrum shaper.

A music professional tries out Caddco's three-channel console.

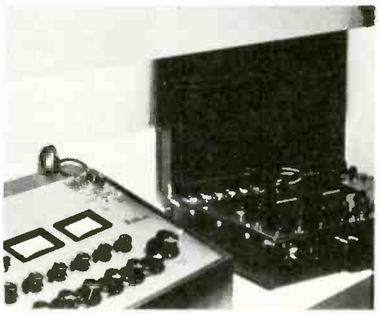
Super-size models of cutting styli decorated the booth of Capps & Company, one of the oldest stylus manufacturers in the business.



Scully's third recorder, circa 1920, was probably the first gravity-operated model ever seen by most visitors.



Fairchild exhibits a flat-top mixer, with recessed meters.

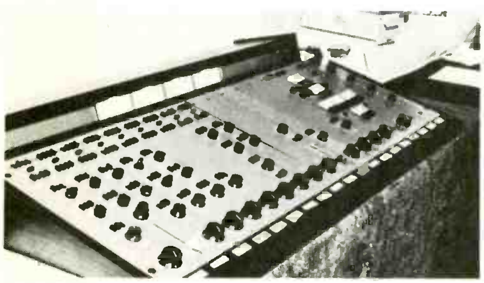


Gately showed two models of its portable mixer.

Rolf Hertenstein of Universal Audio shows off some of their instrumentation.

Custom-built, 8-channel mixer console by Lang Electronics.

Hewlett-Packard's spectacular $8900 Audio Spectrum Analyzer is an engineer's dream instrument.



Langevin showed a flexible and complete console for four channels.

Wiegand Audio Laboratories offers a wide variety of equipment from stock.

Gotham Audio displayed an operating Neumann disc-cutter that cut an LP master from a master tape played on a Studer tape machine.

William G. Dilley, erstwhile contributor to Audio, explains his new Spectra-Sonics solid-state amplifiers.

ABZs of FM

LEONARD FELDMAN

The Ratio Detector

As we saw in last month's installment, the conventional form of FM detector, the Foster-Seeley Discriminator, is sensitive to AM variations as well as desired FM deviation or modulation. As a result, one or more limiter stages is required to remove all amplitude modulation and apply a pure FM signal to the input of the discriminator.

In the mid-1940's, a circuit was developed that was sensitive to frequency variations, though much less sensitive to amplitude variations. This circuit, dubbed the Ratio-Detector, may therefore be thought of as a combination limiter-detector. In widespread use today, the ratio-detector, as used in high-quality tuner designs, is still preceded by one or more limiters. But this is done to afford even more AM rejection than would be possible if the same amount of limiting circuitry preceded a conventional discriminator.

In order to understand the limiting action of a ratio-detector, let us refer to Fig. 1, which is the schematic of the Foster-Seeley discriminator that appeared in identical form in last month's discussion. As shown then, equal voltages appear across R1 and R2 when the incoming i.f. signal was centered at 10.7 MHz (no audio information being transmitted). Assume that with a certain carrier strength and a given modulation the voltage across R1 increases from a quiescent value of 3 volts to 4 volts, and that the voltage across R2 decreases from its static value of 3 volts to 2 volts. The net output voltage (instantaneous) would be the difference, or 2 volts. Now assume a stronger incoming i.f. signal, arising from a stronger carrier signal so that the voltage across R1 and R2 is now 6 volts instead of 3 volts. With the same modulation applied as before, but with this stronger carrier, the voltage across R1 will rise to 8 volts while the voltage across R2 will decrease to 4 volts, a net difference (and instantaneous output) of 4 volts.

Despite the fact that the same modulation has been applied in both cases, the amplitude of output in the second case is twice as large as in the case of the weaker incoming r.f. signal. This amounts to direct AM response as well as FM response and is the reason why limiting is needed with the discriminator circuit. There is, however, an interesting numerical relationship between the two examples cited. The ratio of voltages across R1 and R2 in the first example, 4/2, exactly equals the ratio of voltages across these resistors in the second example, 8/4. It is this equality of voltage ratios, regardless of incoming signal strength, that gives rise to the idea of the ratio-detector.

The concept of ratio-detector operation will be best understood by examining the circuit of Fig. 2, in which each diode is associated with a completely separate resonant circuit. Let us again assume that the circuit involving D1 is tuned above 10.7 MHz, while that of D2 is tuned below 10.7 MHz. The output voltage for the D1 circuit will appear across C1 while the output voltage for D2 will appear across C2. The battery V_b is a fixed voltage. Since C1 and C2 in series are directly across this battery, the instantaneous sum of their voltages must equal V_b. Also note that the polarity of the battery is such that under static or no signal conditions, no current can flow around the circuit consisting of T1, T2, D2, C2, C1, D1 and T1. Now, although E1 + E2 can never exceed or be less than V_b, E1 does not necessarily have to equal E2.

In other words, the *ratio* E1/E2 may vary! Output signal (recovered audio) can be taken from a variable load resistor connected across C2, since the voltage across this capacitor will vary with signal.

When the incoming signal is unmodulated (10.7 MHz), E1 and E2 will be equal. This is similar to the situation that prevails with the discriminators examined earlier. When the incoming signal rises in frequency because of modulation applied, it approaches the resonant frequency of T1, resulting in a higher voltage across C1. At the same time, a lower voltage is developed across T2. Therefore the voltage across C2 decreases. However, the sum of these two voltages must still equal V_b.

In other words, an instantaneous change in frequency alters the ratio of E1/E2, but not the total voltage. With a signal frequency modulated to a point below 10.7 MHz, E2 will exceed E1, but, again, the ratio of E1/E2 will remain constant because of the stabilizing effect of V_b. Desired audio information is obtained from the voltage variations across C2. Since only audio variations are desired, C3 serves to block the d.c. voltage present at all times across C2.

The key to this explanation is, of course, V_b, which keeps the total voltage constant while permitting the ratio of E1/E2 to vary. Thus, in this elementary circuit, an output voltage is obtained that is purely a result of the

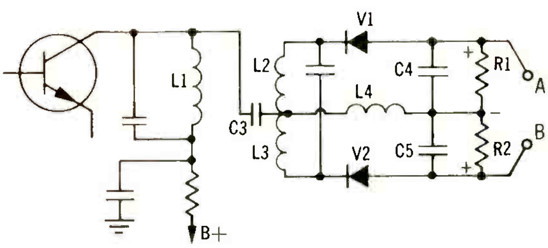


Fig. 1 – The Foster-Seeley discriminator.

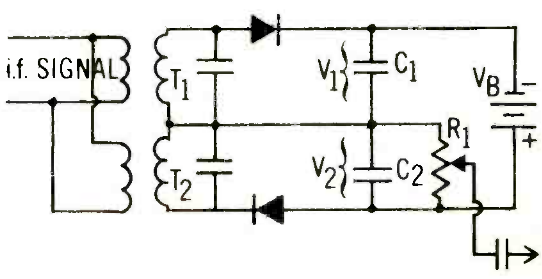
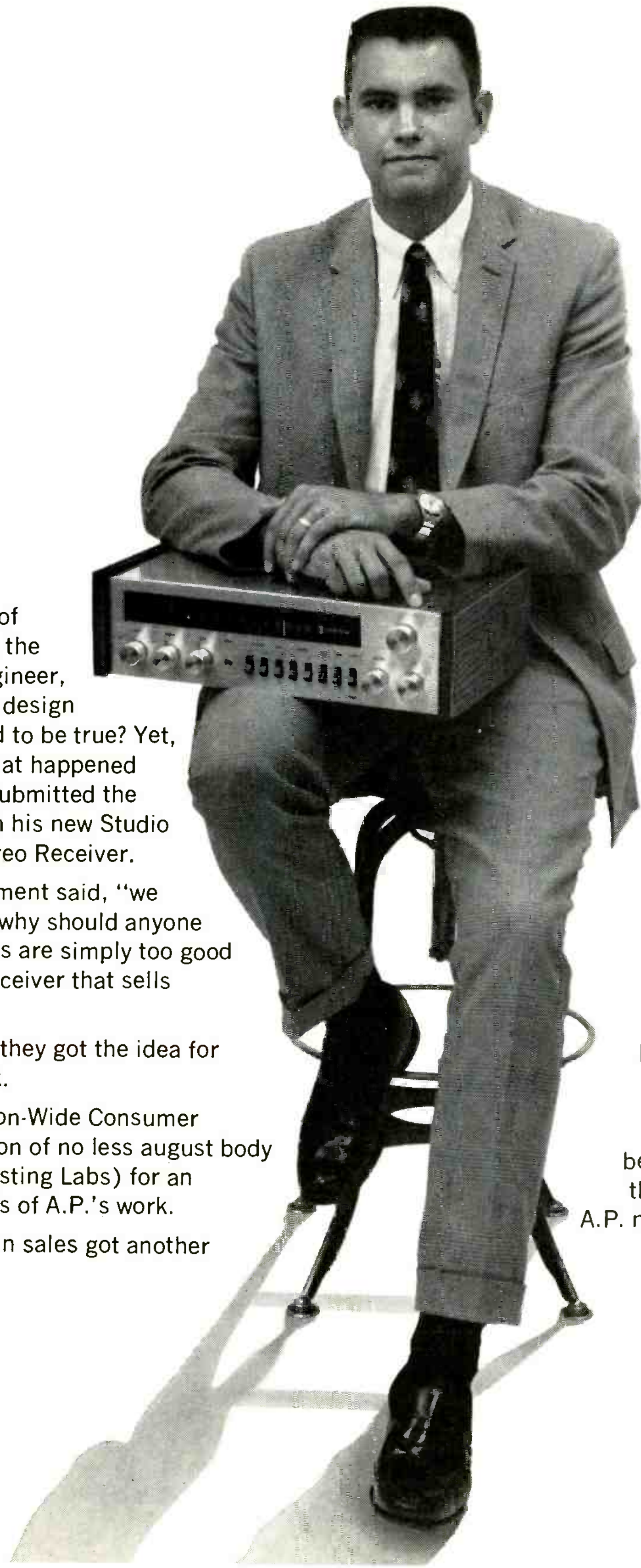


Fig. 2 – Explanatory form of ratio detector.

Our A.P. Van Meter designed the PRO-120 so well, he had to go through the indignity of being double checked.



Who ever heard of double checking the head project engineer, just because his design seemed too good to be true? Yet, that's exactly what happened when A.P. first submitted the specifications on his new Studio Pro-120 FM Stereo Receiver.

"A.P.," management said, "we believe you, but why should anyone else? These specs are simply too good to be true in a receiver that sells for $379.50!"

And that's when they got the idea for the double check.

They called Nation-Wide Consumer Testing (a division of no less august body than the U.S. Testing Labs) for an impartial analysis of A.P.'s work.

Then somebody in sales got another bright idea, "Why not ask them to certify that the Pro-120 will meet or exceed its published specifications?"

The men from U.S. Testing agreed, but on one condition. They wouldn't test a Pro-120 at their lab. (After all, anyone who cares can "tune-up" a unit just as you would a car.) Instead, they would come to University in Oklahoma City and pull units at random right off the production lines.

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FM signal. In actual practice, the use of a battery would severely limit the dynamic range of such a circuit. For example, if V_b were made very high, weak incoming signals would be lost entirely because they would not possess sufficient amplitude to overcome the negative "back bias" placed on diode D1 and D2 by V_b. If V_b were to be made quite low, then more powerful stations would be severely limited in the amount of audio voltage that could be recovered from the circuit because the voltage across either capacitor (or even the sum of voltages across both C1 and C2) could never exceed the low value imposed by V_b.

In practical circuits, the value of V_b is not determined by a fixed battery, but by the average value of each incoming carrier. This idea will be understood better by examining Fig. 3, a practical form of ratio detector.

This circuit uses the same phase-shifting properties as the discriminator of Fig. 1, which was fully explained last month. R and C3 replace V_b, the battery, and the voltage developed across R will be dependent upon the strength of incoming signal. Notice that D1 and D2 are in series with R and all current flowing through these diodes must flow through R. By placing a 5-μf capacitor across R, a fairly constant voltage is maintained and momentary changes in signal amplitude are absorbed by this capacitor. It is only when the average value of the incoming signal changes (as when tuning from a stronger station to a weaker one) that the voltage across R changes significantly. Audio output is still taken from across C2. Since the voltage across R is dependent upon incoming signal strength, it may be used as an AGC (automatic gain control) voltage. The ratio detector in this form does possess the disadvantage of being somewhat more difficult to align; also, great care must be taken to obtain a linear response characteristic both during alignment, as well as in the design of the ratio-detector transformer in the first place.

A more symmetrical form of ratio-detector design (and one that is almost always used in preference to the form just discussed) is shown in Fig. 4. Instead of the direct capacitive connection between L1 and the secondary tap, we now introduce L4. Using the same reasoning as was applied in our earlier discussion of discriminators, the voltage induced in L4 (which is closely coupled physically to L1) will remain constant so long as the primary voltage is constant. Since the voltage across L4 depends upon the voltage across L1,

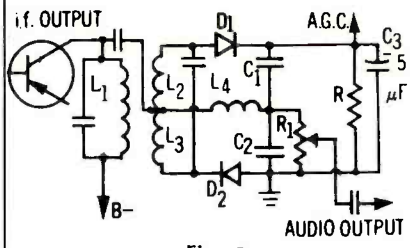

Fig. 3

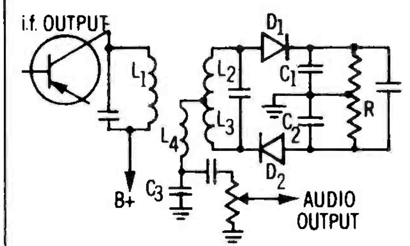

Fig. 4

Fig. 3 – Practical form of ratio-detector circuit.

Fig. 4—Balanced or symmetrical ratio detector. (Note: "R" may consist of two fixed resistors in practice.)

and not upon the secondary circuit, it can be used as the reference voltage in place of the previous capacitive-coupled arrangement of Fig. 3.

The path from the center tap of the secondary to the junction of C1 and C2 includes L4 and C3. Voltage applied to D1 consists of E_{L4} and E_{L2}, while voltage applied to D2 consists of E_{L4} and E_{L3}. At 10.7 MHz (center i.f. frequency), both diodes receive the same voltage, with C1 and C2 reaching the same potential. The current in D2 flows up through L3, through L4 to C3 to ground, and thence through C2. Current continues to flow until the voltage across C3 and C2 equals the voltage at the diode. Current from D1 flows from its cathode through C1, C3, and L4 and up through L2, and back through D1 until C1 and C3 are charged to the voltage at the diode. The current from D2 flows through L4 and C3 in the opposite direction from the current produced by D1. Therefore, these two currents in the common branch (L4 and C3) actually oppose each other and the resultant voltage is zero.

At frequencies above and below 10.7 MHz, the voltage applied to each diode will vary and a net current will flow through C3 and L4. When frequency varies in one direction, a positive voltage will be developed across C3 and L4; when frequency varies in the other direction, a negative voltage will be developed. The potential variations across C3 will vary directly with frequency and, therefore, represent audio information.

Both the ratio detector and the discriminator have one advantage in common in that there is a point in each circuit at which there is a zero d.c. potential when the set is tuned accurately to the center of the channel. First, the popular zero-center tuning meter can be connected (with suitable impedance isolation) to this circuit point. When a station is properly tuned in, the meter pointer will rest at the center of the scale. When the station is detuned, the needle will indicate a positive or negative voltage, depending on which way the signal is detuned. This type of meter is the easiest to use, visually, and is often present in good-quality tuners and/or receivers.

Perhaps more important, the "zero center" feature of these circuits provides a convenient take-off point for some form of AFC (automatic frequency control) used to "lock in" FM stations, once properly tuned. Without detailing these AFC circuits at this time, it is obvious that a voltage that is zero when a station is properly tuned in and varies above and below zero when the station is detuned can be used to retune the receiver electronically.

To sum up the difference between the ratio detector and the discriminator, the latter operates on the difference of the output voltages of two diode detectors. Diode load resistors are connected with their voltages in opposition. The resultant of the two voltages becomes the audio output voltage. Since the discriminator responds to amplitude modulation as well as FM, it must be preceded by one or more limiters and requires a great deal of amplification in order to drive these limiters into saturation. Ordinarily, this requirement makes for a more costly design.

In the ratio-detector circuit, the two diodes are connected in series and a stabilizing or controlling voltage is developed that depends upon the strength of the incoming carrier signal. The controlling voltage sets the limit of maximum audio voltage that can be obtained. The ratio-detector is more immune to AM variations than the Foster-Seeley discriminator (considered without limiter). And it is generally more economical since it requires fewer amplification stages ahead of it. Often, a limiter, or an i.f. stage acting as a partial limiter, will precede a ratio detector to improve AM rejection further. As a consequence of the foregoing, most "hi-fi" tuners today employ a ratio detector. Æ

20 QUESTIONS

(Continued from page 34)

word of caution! Unless your TV set is fully transformer-powered (chassis not connected to one side of the power line in any way) the above connection should not be attempted, as a shock hazard may result. Many small portables, unfortunately, are not "transformer powered," and so the foregoing would not be advisable for use with such "hot" sets.

Q. *As of now, is solid-state (transistorized) equipment as "perfected" and as reliable as "good old" tube equipment?*

A. Yes, and in many ways it's better. The hi-fi component industry (like everyone else) rushed headlong into transistorization—too early in the game at that. As a result, some early pioneer users are justly soured because of some of the early failures. Germanium output transistors were notoriously sensitive to overheating, to the point of self-destruction. These have been largely replaced by more rugged silicon devices, along with fused or automatically protected output circuits. Conventional bi-polar transistors used in the r.f. section of most early tuners really weren't ideal for the job, but nothing else was available, so selectivity and cross-modulation specifications suffered in early models. Today, FET's (Field Effect Transistors) practically do as good a job in this area as the best r.f. triode vacuum tubes were able to do in the "old days." Add to this the longevity of transistors as opposed to tubes, as well as the reduction in component-destructive heat, and we must conclude that transistorized equipment *today* is without doubt preferred over vacuum-tube hi-fi equipment.

Even the cost factor has done a turnabout. In the past we might have had to say that solid-state hi-fi is great if you want to spend the extra money. Today, you're probably ahead even financially when you buy solid-state high fidelity components designed and produced by the many reliable manufacturers who are members of our industry.

Well, that concludes our question and answer session! What? You just counted up the number of questions and find that there are only nineteen? Oh, yes, the twentieth question most often asked was:

Q. *Why don't you publish some of these questions and answers for us newcomers to the field?*

A. We just did!

AUDIO MUSIC REVIEW

Light Listening

STUART TRIFF

The Fall Harvest

JUST AS SURE as Autumn brings the falling leaves, so, too, an avalanche of new record releases is certain to come tumbling down around our heads at this time of the year. In an effort to give due attention to the most important and worthwhile entries in time for Christmas gift giving, and before the new year is upon us, herewith is a roundup of the pick of the crop.

Theatre & Film

Heading the list are the soundtracks of the film versions of "Funny Girl" (Columbia BOS-3220) and "Finian's Rainbow" (Warner / 7-Arts BS-2550). In the former, several numbers from the original stage production are dropped in favor of two new ones written especially for the film, plus a couple of Fanny Brice oldies. No matter, BARBRA STREISAND is still the whole show, as she was before, but now she brings an even deeper emotional maturity to her part. OMAR SHARIF as Nicky Arnstein, is fine in his brief singing chores, and the recording has a warm, well-rounded sound—much better than the usual soundtrack job. "Finian's Rainbow," I'm afraid, is something else again. We've waited more than twenty years for the celluloid treatment of this one, only to find it exceedingly miscast, save for PETULA CLARK, who manages to bring some semblance of style to the proceedings. FRED ASTAIRE has little to do vocally; TOMMY STEELE hams it up outrageously; and DON FRANCKS sounds just plain anemic in the role of the husky sharecropper.

"Star!," (20th Century Fox DTCS-5102), the lavish film musical based on the life of the incomparable Gertrude Lawrence, has the considerable advantage of JULIE ANDREWS bringing luster to a couple of dozen wonderful tunes ranging from "Limehouse Blues" to "My Ship" . . . a delight from beginning to end! A word of praise is also in order for Lennie Hayton's stylish arrangements and musical direction.

The soundtrack of "Chitty Chitty Bang Bang" (United Artists UAS-5188), based on Ian Fleming's story about a magical car, has songs by the Sherman brothers, very much in the same vein as their "Mary Poppins" score: DICK VAN DYCK and SALLY ANN HOWES are ingratiating performers. The kiddies should love it, but it might be pretty gooey going for adults.

An orchestral rendering of "The Happy Time" (RCA Victor LSP-3986), strengthens my already high opinion of John Kander's romantic and tuneful score. It's played by a crackerjack group billed as "Orchestra '70," led by JOE REISMAN, who, along with Marty Manning, did the fresh and imaginative orchestrations. Top-notch stereo.

Instrumental

The two most exciting and influential talents in pop music to come along since Michel Legrand first burst on the scene some years ago, are England's LES REED and France's PAUL MAURIAT. In his latest album, "New Dimensions" (Deram DES-18011), Reed is featured in the multiple roles of composer-arranger-conductor-pianist in fine-sounding versions of his own "Misty Morning Eyes," "The Last Waltz," and "There's a Kind of Hush." The latter also serves as the title of a collection of Reed's tunes with the GEOFFREY EVANS ORCHESTRA (Palette PTS-30,000) playing lush, but swinging arrangements by ARTHUR GREENSLADE.

"Mauriat Magic" (Philips PHS-600-270) aptly describes the maestro's unique talents for off-beat scoring, as he leads his orchestra in a highly enjoyable program of some of today's better pop tunes: "Love in Every Room," "When You Get to San Francisco," "Live for Life," and almost a dozen others. A beautifully realized recording that should wear well with repeated hearings. The album photo, by the way, reveals a young lady who knows how to wear very little very well.

Paul Mauriat was responsible for the great success of "Love Is Blue," in behalf of his confrère, ANDRE POPP, who conducts a collection of his own songs (MGM SE-4564), spotlighting ear-popping renditions of such lilting originals as "Why Say Goodbye," "Tililoy," "Bim Bom," and, naturally, "Love Is Blue."

RAY CONNIFF, in "Turn Around and Look at Me" (Col. CS-9712), directs his excellent group of singers and instrumentalists in driving, somewhat tedious arrangements of "Mrs. Robinson," "MacArthur Park," and "Angel of the Morning."

Pop Vocals

Bowing to the protocol of ladies first, we have at hand new discs by some of the best in the business. NANCY AMES, recorded "live" at New York's Hotel Americana (Epic BN-26378), runs the gamut from Beatles to bossa nova, from "Michelle" to an exciting Latin-American medley, complemented by Joe Sherman's fine settings. Contrasted with Miss Ames' electric, swinging style, is the soft, intimate and caressing quality of ASTRUD GILBERTO in "Windy" (Verve V6-8754). Brazilian songs predominate here, with Luis Bonfa's "Dreamy" and arranger-conductor Eumir Deodato's "On My Mind" particularly outstanding.

The smoky warbling of LANA CANTRELL is displayed to good effect in "Lana!" (RCA Vic. LSP-4026), but excessive use of an echo chamber tends to become annoying after a while. Nicely-varied selections include "Gentle on My Mind," "The Fool on the Hill," and "Can't Take My Eyes Off You."

The latest by PETULA CLARK (Warner / 7-Arts WS-1743), is devoted principally to songs by her favorite composer-arranger, Tony Hatch. My own picks, among those by the talented Mr. T., are "Don't Give Up," "We're Falling in Love Again," and "Beautiful in the Rain."

Next to Barbra Streisand, whom she once understudied, LAINIE KAZAN is unquestionably the most exciting female singer of the decade! Her voice is extraordinary, with phenomenal range and color, uncommon among pop vocalists; and the lady really savvys every nuance of every lyric she sings. All this is put forth in abundance on "Love Is Lainie" (MGM SE-4496), in which Miss Kazan delivers emotion-packed, touching interpretations of "When I Look in Your Eyes," "Sunny," "How Can I Be Sure," and "The Look of Love." She makes everything sound better than it really is, and I know of no higher praise to bestow on any singer.

Leading off the male contingent is ROBERT GOULET in "My Love Forgive Me" (Col. CS-9096), arranged and conducted by Ralph Burns. Though somewhat lacking in personality, there's otherwise no faulting this well-chosen program of easy listening stylings of "Softly, As I Leave You," "Corcovado," and "This Is All I Ask." ED AMES, on the other hand, need not "Apologize" (RCA Vic. LSP-4028) for the solid baritone and individual style he brings to "Born Free," "Honey," and "Scarborough Fair."

TONY BENNETT, an ever-busy man in the recording studio, comes up with a nice album of standards on "Yesterday I Heard the Rain" (Col. CS-9678). The best tracks are "Love Is Here to Stay," "Get Happy," and "I Only Have Eyes for You"; all enhanced by arrangements by the gifted Torrie Zito.

Christmas

Three new Christmas albums stand head and shoulders above all the rest. "The Story of Christmas" (Col. CS-9738), featuring 19th Century music boxes from the collection of antique dealer Rita Ford, is an absolute charmer. The strange, yet serene quality of these incredibly varied-sounding instruments playing carols, makes for a most unusual holiday record. Traditional Carols, in quite a different approach, are put forth by RAYMOND LEFEVRE and his Orchestra (4 Corners FCS-4257). These are contemporary pop settings that still manage to be tasteful, fresh, and entirely reverent. BURL IVES' "Christmas Album" (Col. CS-9728) offers the famous ballad singer's interpretations of a dozen carols that are as heartwarming as a cozy hearth on a cold winter's night . . . highly recommended!

HI-FI REFERENCE LIBRARY

(Continued from page 18)

Blue Ridge Summit, Pa., 104 pages, $6.95, hardbound; $4.95, soft cover, uses a "pictured' 'text" approach to show readers how to use an oscilloscope. Large drawings and basic operating procedures through to clearly written and illustrated examples of waveform displays makes this a fine guide book for anyone who wants to learn how to utilize a 'scope. **How to Use Signal Generators in Radio/TV/Hi-Fi Servicing** by John D. Lenk. John F. Rider Publisher, Inc., New York, N. Y., 120 pages, $3.25, examines the use of various signal generators — including audio generators and FM stereo generators. In addition to discussing basic operating procedures, step-by-step details on how to test an antenna system, AM receiver, and FM receiver, basic amplifier tests are given, as well as test equipment calibration information.

And let us not overlook the helpful source material listings published by **Schwann Long Playing Record Catalog** (Monthly, 45¢), the annual **Polar Index to Record Reviews**, 20115 Goulburn Ave., Detroit, Mich. ($2.00), an invaluable aid to locating record reviews published in 11 national magazines, and the quarterly **Music Article Guide**, 156 W. Chelten Ave., Philadelphia, Pa. ($12.00 per year), which lists feature articles on music that appeared in American periodicals. Æ

Classical Record Reviews

EDWARD TATNALL CANBY

Haydn: Double Concerto in F for Violin, Harpsichord and Orchestra; Symphony No. 44 in E Mi. ("Trauer"). J.-F. Manzone, Vl., Françoise Petit, Harps., Henri-Claude Fantapié and His Chamber Orch.
Baroque 2867 Stereo ($5.79)

One after another, hundreds of Haydn works are being restored to currency, the old sources collated and corrected, newly published editions played and in due time recorded for our continued pleasure. The quantity is nearly endless, and the quality of the music unfailingly high. Here is an alleged "premiere performance" (that is, since the eighteenth century) of one of those rather special concertos of the Mozart-Haydn period often called *sinfonia concertante,* featuring more than one solo instrument, which became immensely popular in the late 1770s and has never since lost its appeal. (Even Brahms wrote one, in his own style.)

Being early Haydn, this one is light and airy, full of graceful melody and ornamentation—not a heavy sound in it. The playing is excellent, with a sensitively attuned string orchestra and two fine soloists. The recording, too, is beautifully done, the tricky balance between solo violin, the faint solo harpsichord and the small orchestra precisely right for records. In particular, note that the harpsichord — which can so easily be "blown up" to gigantic size, is kept exactly in the right proportion as a faint, silvery sound blending in with the strings and yet clearly expressive as a solo, in its own right. Somebody knew his business.

The "Trauer" Symphony is one of the handful in a minor key out of hundreds. Coming just before the more familiar "Farewell," No. 45, it will quickly remind many listeners of that superb work though its music is somewhat fiercer and more "trauer" (sad), as per the title.

Performances: A— *Sound:* A—

Haydn: Overture to an English Opera; Symphony No. 63 "La Roxolane"; Symphony No. 78. Little Orch. of London. Leslie Jones.
Nonesuch H 71197 Stereo ($2.50)

The Leslie Jones series of Haydn recordings on Nonesuch now has reached impressive numerical proportions as well as artistic, though if I am right they have not all been made in the same place nor by the same engineers. There is thus a certain variance in the sound; in this one the reverberation time is short, the ensemble a bit bass-heavy (note the very prominent drum beat) and very slightly strained—undoubtedly a matter of hall acoustics and mike placement. In minutes, the ear adjusts, and then forgets.

The English Overture, brief and brilliant, is a very late work from Haydn's English period. Nothing big or important, but a pleasure to hear. (It seems a trace under-rehearsed; a few phrases are uncertain in sound.) "La Roxolane" (much more secure in the playing) is a *pastiche* symphony made of retrieved parts of other works—no matter! It is one of the most forceful of the middle-period symphonies, beautifully projected by the Jones players. No. 78, one of the few in a minor key, reminds us at once of No. 44, the "Trauer" but even more of Mozart. It is a rich, elaborate work, already in the last-period manner, with a profusion of clear-cut tunes and themes, immensely vigorous in its harmonic developments, and ending with a typically tuneful rondo finale. On casual hearing, you could well mistake this—until the last movement—for a late Mozart symphony. Over a period of some years the two composers interacted remarkably, directly affecting each other in the most profound manner.

The slightly different sound aside, this is one of the finest of the Jones Haydn records.

Performance: B+ *Sound:* B

Piano-Roll Blues

George Gershwin Plays Rhapsody in Blue and Other Favorites. (Duo-Art Piano Roll.)
Everest X-914 Stereo

Though it is impossible to review all of the dozens of discs in this piano roll series, this Gershwin performance is not to be passed by. How did the master of early pop music sound in his own right?

Well—funny. Wouldn't have believed it. The familiar Rhapsody, of course, is an orchestral work, not a piano piece; but Gershwin was a pianist and the orchestrating of his big works was done by Ferde Grofe, an expert. Gershwin didn't know how. Thus we can assume that this piano version is the original, or a version thereof. Authentic as all get-out.

How does it sound! Well, it races. So fast you gasp for air. He really rips along. Not continuously—he is still "Romantic" enough to slow down, again and again, for that poignant effect . . . but when the themes take over, they whip along, almost too fast for the ear. Probably authentic, too, for the player piano super-rolls at least do reproduce the correct tempi, as played, assuming that the mechanism is working right.

The second side is odd. These are typical "player piano" versions of popular tunes, massively tricked up with incredible riffles and slides and side-effects, a sort of updated and heavyweight version of the older music box sort of arrangement. I found these totally commercial items perfectly delightful — what marvelous playing and what a superbly saucy piano sound! Even on a grand, one of those aristocratic 9-foot Steinways. I ask only one question—could ten fingers, just ten, play all these billions of notes???? These four short items sound to me like piano *duets,* for two performers and twenty fingers. If in truth Gershwin played all the notes himself, then he was indeed some pianist. (But Everest could have made a slight boo boo. Wouldn't be their first.)

Performances: A *Sound:* A—

Important P.S. The Everest piano rolls come in two series. If it doesn't say "Series II" on it—forget it. The first set was pretty awful. But they fixed the machinery up for the new series, and now it works fine. All the good ones are marked Series II.

Pierre Boulez Conducts Berlioz Symphonie Fantastique, Lélio. Jean-Louis Barrault, Narr., John Mitchinson, Ten., J. Shirley-Quirck, Bar., London Symphony Orch. and Chorus.
CBS 32 B1 0100 (2) Stereo ($7.79)

Now if you want *length*, and are tired of Mahler and, maybe, Wagner, then try a bit of early Berlioz. Phew!

The familiar and lengthy "Fantastic Symphony" expressed the composer's vast-scale fantasies concerning his love, one Harriet Smithson; the poet's soul is represented by a musical *idee fixe*, a mournful and somewhat vapid theme that appears in all the movements. But the "Fantastic" wasn't big enough for Berlioz. He later added another monster work on the same subject, "Lélio," and ordained that it should be heard and seen immediately after the "Fantastic." It seldom has been; but on these discs it is. Heard, anyhow.

In Lélio, the soulful poet appears in the flesh and spouts French, great volumes of it, with an occasional break for a huge musical set piece, vocal solo, piano, orchestra, large chorus. There's even a slightly incredible play within a play, a fantasy on a scene from Shakespeare's "Tempest" (Harriet Smithson was a Shakespearean actress).

Just to make the whole double bill ultra-clear(?), Berlioz begins and ends "Lélio" with the familiar "Fantastique" theme. (At the very end, the speaking voice of Lélio, as the theme plays, is heard to mutter, "Encore, et pour toujours!," which I am moved to mis-translate, "That d——— theme again! Let me out of here!! Forever.")

It's all very much of the early-Romantic avant-garde movement, which is fine if you are tuned to that sort of thing. Otherwise, you will find it mostly a vast gush of emotional hot air, first one desperate, cliché-ridden mood, then another, all described in anguished detail. There are, as always, magical moments in the music, but also plenty of pretentious musical hot air, too. That was the youthful Berlioz all over. (Later, he had more to say, if at no less length.)

A fine performance of the whole double feature, the big chorus particularly good in the "Lélio" climaxes.

Performances: B+ *Sound:* B+

Ravel: Schéhérazade. Berlioz: Les Nuits d'eté. Janet Baker, soprano; New Philharmonia, Barbirolli.
Angel S 36505 Stereo ($5.79)

Janet Baker is so very English that it is a double pleasure to hear her clear, true voice in an excellent approximation of French singing style. That is a feat which few Britishers can manage, nor, for that matter, Americans. French music has its own sound and nowhere more so than in the special French voice, unlike any other.

These two lovely works for solo soprano and orchestra are both from young composers, Berlioz in 1934 (but orchestrated later), Ravel around 1903. They are oddly contrasted. Berlioz, as always, is full of deceptive energy but his early-period harmonies are almost over-simple, only the peculiarly free modulations from key to key giving away to the modern ear their once-radical impact. Ravel, master of the most complex color-harmonies, writes here at his exotic and Impressionist best, the music laden with color effects of every sort.

Janet Baker is superb in the Berlioz, her own directness of approach similarly concealing a great deal more art than appears on first hearing. In Ravel, she is outclassed only by such masters (mistresses) of vocal color as Jennie Tourel, who once did a marvelous recording of these exotic and lush works with our own exotic Leonard Bernstein (still available on Columbia MS 6438). But Baker's version is so strong in its own way that perhaps "outclassed" is the wrong word. Tourel merely sounds more like a French singer. Both are excellent versions.

And never forget Maggie Teyte, the ultimate French singer, who was born pure British! In mono, you can still hear her in an Angel COLH reissue, No. 138, made many years ago. Then there's De los Angeles (Spanish), and Crespin . . . I could keep this up for pages.

Performances: A— *Sound:* B+

Prokofieff: Classical Symphony, Op. 25 (1916-17); Symphony No. 7 (1952). Moscow Radio Symphony Orchestra, Rozhdestvensky.
Melodiya Angel SR 40061 Stereo ($5.79)

This Russian performance of the familiar "Classical" Symphony is a pleasing revelation. After umteen dozen smart-aleck virtuoso performances by American and European orchestras, who seem to think the thing to do is to play the music twice as fast as the competition, like a sort of orchestral finger exercise, this version at last moves at a wholly reasonable speed throughout. And so we hear once again the many lovely "classical" (i.e. Mozart-Beethoven period) details, so nicely turned out, with such affection, such warmth and fantasy! This is the way it ought to be, this little symphony.

The much later Seventh, full of nostalgic warmth too, blending the Prokofieff brand of lyric humanity with the new postwar Russian manner, boldly mechanistic and hard, gets only a so-so realization. Somehow, the telling motives, the memorable harmonic twists, the poignant moments, are treated too casually, if accurately. Doesn't really "emote," this performance. But the disc is worth it for the "Classical."

Performances: A–, B– *Sound:* B–

Kurt Weill: Symphonies No. 1 (1921), No. 2 (1933-34). B.B.C. Symphony Orch., Gary Bertini.
Angel 36506 Stereo ($5.79)

An unusually interesting discovery, these two virtually unknown "serious" symphonies by the famed pop composer of "Threepenny Opera," "Lady in the Dark" and many another dramatic work of the twenties and thirties. They are surprisingly good symphonies, though flawed here and there: we can forget about the pops angle, for Weill turns out to have been a very professional writer for his classical orchestra. He was no Gershwin (who had Grofé do the work for him). Most interesting of all is that these highly "classical" works make no concessions whatsoever to popular music and yet manage to sound unmistakably like Kurt Weill, even so, for all of us who have heard his popular theatre music.

Symphony No. 1, dating from 1921 (the composer was 21 and just beginning) is, of course, extremely dissonant. All serious music had to be dissonant in those days. You couldn't get away with an ordinary chord. But there are clear Weill-like tunes, even here. And, as well as in No. 2, those long sequences of endless four-measure phrases, one after the other, with hypnotically simple rhythmic accompaniments, that are so typical of Weill, the stage composer. In spite of the dissonance, there are youthful touches of Schoenberg, Strauss, even Brahms—Weill was still basically a student.

No. 2, written when consonance and old fashioned keys were coming back into style, is a slicker and more nearly popular work though still masterfully put together as a symphony. You can almost hear Lotte Lenya's voice in passage after passage. Amazing. The performances are astonishingly good, full of understanding and imagination.

Performance: A *Sound:* B+

C. P. E. Bach: Double Concerto in F for Two Harpsichords; Concerto in B Flat for Cello. Gustav Leonhardt and Alan Curtis, Hps., Angelica May, Cello, Collegium Aureum Orch.
RCA Victrola VICS 1342 Stereo ($2.50)

Carl Philipp Emanuel Bach, middle son of the three musical sons of old J. S. Bach, is at last coming into his own on records, and deserves it. C. P. E. Bach is more inward, more thoughtful than his somewhat brittle and fashionable younger brother, Christian Bach, and yet — though C. P. E. often reminds us of old J. S. himself—there is already a grace and polish to the sound that begins to point to Mozart. A nice combination of style, and he is a good composer

The two harpsichords purr silkily in this excellent and well balanced recording, never too loud in comparison with the orchestral sound. The solo cello, too, blends remarkably easily into the orchestral backing, without a trace of those unpleasant groans and squawks that we usually associate with virtuoso cello playing. Like the harpsichords, the cello is given a proper musical balance. Good.

Performances: A– *Sound:* B+

A Recording First! Glenn Gould Plays Beethoven's 5th. Transcribed for Piano by Franz Liszt.
Columbia MS 7095 Stereo ($5.79)

Maybe it's a recording first, but a zillion good pianists have played the Beethoven Fifth, or portions thereof, on a zillion pianos; and so I find this version slightly less than sensational, all things considered. Now if only Gould had put away the old-fashioned and outdated Liszt arrangement and just read the music off directly from the orchestral score. Or played it by memory. Plenty of good pianist musicians can do that and do it very convincingly.

In other words — Glen Gould is a splendid musician but look, Columbia, even *he* can Do Wrong. He did it.

Performance: B *Sound:* B–

Ars Nova.
Elektra EKS 74020 Stereo ($4.79)

What happens to drop-outs from our classical music conservatories—say, the Mannes Schoal in New York? Some of them drop music for something more remunerative, like toothpaste. Others start a Group.

Ars Nova, named after a "new art" that flourished in the fourteenth century, shows what comes out. These fellows managed to absorb a lot of classical stuff, before they got fed up with school or fired, whichever it may have been. Music of many periods and styles. So they build their pop music on what they'd heard, like anybody else. Why not? And they do a lot better than old Freddie Martin, when he "jazzed the classics" a generation ago

In this pops music you will hear—if your ear is quick and knowing—an astonishing potpourri of musical quotes and parallels, ranging all the way from the old Ars Nova itself through "Zarathustra" by Richard Strauss. There's even a neat sixteenth century Pavan ("Pavan for My Lady"), though whether it is real or composed by the boys in the band I don't happen to know. Or care. Old instruments, too. And antique sounding interludes between numbers. It's all quite a svelte commentary on conservatory education, and I trust the Mannes School people are blushing.

Classical Ragas of India. Nat. Raga Company of India.
Everest 3217 Electronic Stereo ($4.98)

Here is a somewhat different story. The music is basically true-Indian and the intent is to convey to Western listeners at least a portion of the ancient traditions of this music. But something odd goes on here. There are eight "ragas" performed on this disc, the longest only 5:30 and the shortest 2:11. Except for one, occupying most of side 2, "Transcendental Meditations on the Mystique of Spiritual Love," which lasts 22:39. Now that is a reasonable length for a raga, considering that most of the real (non-recorded) ragas go on for *much* longer, and often for hours on end.

Do I detect an understandable desire to reduce the raga to U. S. pop proportions? Maybe to sell it to those who expect their music, Indian or what-have-you, to arrive in the standard three-minute doses? U. S. kids love ragas today. In short lengths, that is.

Performances: ? *Sound:* B–

Jazz

BERTRAM STANLEIGH

Elvin Jones & Richard Davis: Heavy Sounds
Impulse Stereo A-9160 ($5.98)

Among the most sensitive of the new music exponents are bassist Richard Davis and drummer Elvin Jones. With Billy Greene, piano, and Frank Foster, tenor, they produce a disc that is not nearly as heavy as its title suggests. It's serious music making, indeed, but the delicacy and exquisitie detail in the performances is far from heavy. For a label that has always produced top-quality jazz sound, this platter is a new breakthrough toward perfection; the close detail and silent background are super real. I don't remember any brush work or bowed bass passages that have ever sounded so thrilling.

Performance: A *Sound:* A

Johnny Hodges & Earl Hines: Swing's Our Thing
Verve Stereo V6-8732 ($5.79)

Recorded in November of 1967, when both the Ellington band and "Fatha" Hines were playing in San Francisco, this first rate rematch of the Hodges alto and Hines piano makes use of five of Hodges' colleagues in the Ellington ensembles: "Cat" Anderson, trumpet; Sam Woodyard, drums; Jerry Castleman, bass; Buster Cooper. trombone; and Jimmy Hamilton, clarinet and tenor. The title is an excellent clue to the character of the performances. Everyone swings brightly and easily. Clearly this was a labor of love for all concerned, including sound engineer Walt Payne. You won't want to overlook this one!

Performance: A+ *Sound:* A

Toots
Command Stereo RS 930 SD $5.79)
Guitarist Toots Thielmans has often been recorded with musicians like Al Casamenti, Gene Bertoncini, and Bucky Pizzarelli, guitars; Ronnie Zito, drums; and Dick Hyman, piano and

organ. But who ever expected Ron Carter and Herbie Hancock to find their way on to a session of this sort. The results make it clear that Toots is a guitarist of genuine stature when he has ideas of Hancock's calibre to respond to. It's not a profound disc, but it has much more real quality than the expected pop romp, and the Dolby-assisted sound is stunning.

Performance: B *Sound:* A

Gerry Mulligan Quartet: What Is There to Say?
Columbia Odyssey Stereo 32 16 0258 ($2.49)

A reissue on its Jazz Odyssey label of a disc first issued in 1959 under the same title, this Columbia platter features Mulligan with Art Farmer, trumpet; Bill Crow, bass; and Dave Bailey, drums. It offers a far better group of collaborators than the disc noted above, and the result is not merely some isolated solos of merit but the kind of real music making where musicians bounce ideas back and forth and achieve a unified statement. The Farmer-Mulligan duet at the end of *Just in Time* is easily worth the price of the platter, and there's a lot more on the same high level. If you missed this one last time, here's a splendid opportunity to get a truly first class collection of sprightly performances.

Performance: A *Sound:* A

European Arrivals

John Dankworth Orch.: The $1,000,000 Collection
Fontana Stereo SRF-67575 ($4.79)

A British altoist and bandleader, Dankworth has received less attention in the U. S. than his talents and high European reputation warrant. His latest, and most ambitious creation is a ten-movement suite purporting to describe as many paintings. While the art ranges in period from 17th century Flemish to Picasso's Blue Period, the composition and performance style is very much of the present day, incorporating traces of Ellington, Parker, and Gillespie in an elaborate orchestration for reeds and brass of material that frequently dips into 12-tone serialism. It's a pleasantly inconsequential exercise that probably won't even disturb that Flemish landscape painter.

Performance: B *Sound:* A

Archie Shepp in Europe, Vol. 1
Delmark Stereo DS 9409

Recorded in Copenhagen for the Danish Sonnet label late in 1963, this set documents the work of five new jazz musicians at a time when they had not yet made a serious impression on the American scene. But they had already developed a depth of expression and maturity of style that was commanding the respect of Europeans and would, in time, gain them the serious regard of American audiences. For the large group of admirers that Shepp's alto has since commanded here, these recording will serve as a valuable document. With Shepp are Don Cherry, cornet, John Tchicai, alto, Don Moore, bass, and J. C. Moses, drums. The recording was made at live performances at Copenhagen's Jazzhus Montmarte, but the audience is properly subdued, and its end-of-number applause has thankfully been deleted from the tapes wherever possible. For those who find Delmark hard to find locally, the firm is at 7 West Grand Street, Chicago, Illinois 60610.

Performance: B+ *Sound:* B

Recorded Tape Reviews

BERT WHYTE

Mahler/Bernstein

Mahler: Symphony No. 6 in A Minor.
Leonard Bernstein cond. the New York Philharmonic Orch. Columbia M2Q992, **open reel, 4-track stereo, 7½ ips** ($10.95)

Leonard Bernstein is noted for his empathy with Mahler and, although he did not start the Mahler "boom," his frequent programming of the works of this controversial composer has given considerable impetus to the growth of the Mahler "cult." Mr. Bernstein's performances of Mahler have in general received critical accolades, and the 6th Symphony has aroused special enthusiasm. The reasons for this will be quite apparent when you hear this brilliant recording. The conducting is obviously a labor of love and his control of all the diverse and contrasting elements that characterize Mahler is nothing short of miraculous. The first movement, for example, is quite fast paced and aggressive, his heavily accented rhythm and propulsive energy quite exhilarating. The contrast with the Leinsdorf/Boston Symphony recording and its slower, almost "draggy" first-movement tempo is startling. The savage, slashing attacks, the relentless drive, and the tremendous dynamic contrasts of the scherzo and finale that Bernstein elicits from his orchestra are stunning. Equally impressive and appropriate is his loving traversal of the poignantly beautiful andante. The playing of the orchestra is ravishing . . . by far the best work from the Philharmonic in a long time.

For those who have the equipment to do it justice, the recording will furnish some awesome sonorities. Strings, brass, woodwinds . . . all are big, bright and clean. In the Hammerschlag, "the hammer-stroke of fate" which occurs three times in the finale, this is the combined sound of a big, resonant, wooden platform struck by a wooden mallet, and a bass drum of colossal size and impact. The acoustic perspective is broad, but good miking furnishes plenty of orchestral definition. Lateral directivity is just about right, with an excellent phantom center channel. In spite of the great fortes, I dodn't hear one smidgen of overload distortion. Transient response was superb, with the famous Mahler cowbells sounding out nicely in their appointed places in the score. There was some print-through and some mid- and high-frequency crosstalk, but making up for this was the exceptionally low hiss, which incidentally has been a distinguishing characteristic of several other recent Columbia tapes. I wonder if they are using some new duplicating process?

As you have gathered, this recording is a winner . . . a "must" for the Mahler aficionado . . . a revelation for the newly indoctrinated.

Cassette "Pop"

Frank Sinatra: Nice 'n' Easy.
Capitol 4XT1417, Stereo Cassette ($5.95)

Frank Sinatra is not hard to take at any time and this cassette is an especial pleasure for it combines an excellently balanced program . . . with Frankie in particularly good voice . . . with some of the best sound I have heard from this medium. I don't know how significant it is, but this tape was dubbed by the common mandrel system, rather than the usual multi-slave recorder. In any case the sound is fine, with some good highs, good transient response. The level was quite high and hiss fairly low even when the cassette was played at a substantial room-filling volume. Typical Sinatra program . . . "Fools Rush In," "That Old Feeling," "Try a Little Tenderness" and others in the same vein. To sum up: this is one of the few cassettes that can be played through a big, high-quality system at a level of quality approaching that of open-reel pop tapes.

Baja Marimba Band Rides Again.
Ampex/AM AMX5109, Stereo Cassette ($5.95)

Here is another outstanding cassette and further evidence that this medium can furnish a fairly high quality of sound when everything is done right. In this case it means an overall high level on the tape, surprisingly wide frequency response, good transients, low hiss. A big, bright clean sound that enhances one of the Baja Marimba Band's better programs. This unique group is very enjoyable in such as "Brasilia," "More," "Spanish Rose," and others.

Other Recommended Titles:

Bobby Hackett: A Time for Love.
Ampex Cassette PJX55016 ($5.95)
The Sandpipers: Guantanamera.
Ampex Cassette AMX5117 ($5.95)
Ed Ames: Opening Night.
RCA Victor 8 track P8S1316 ($6.95)

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