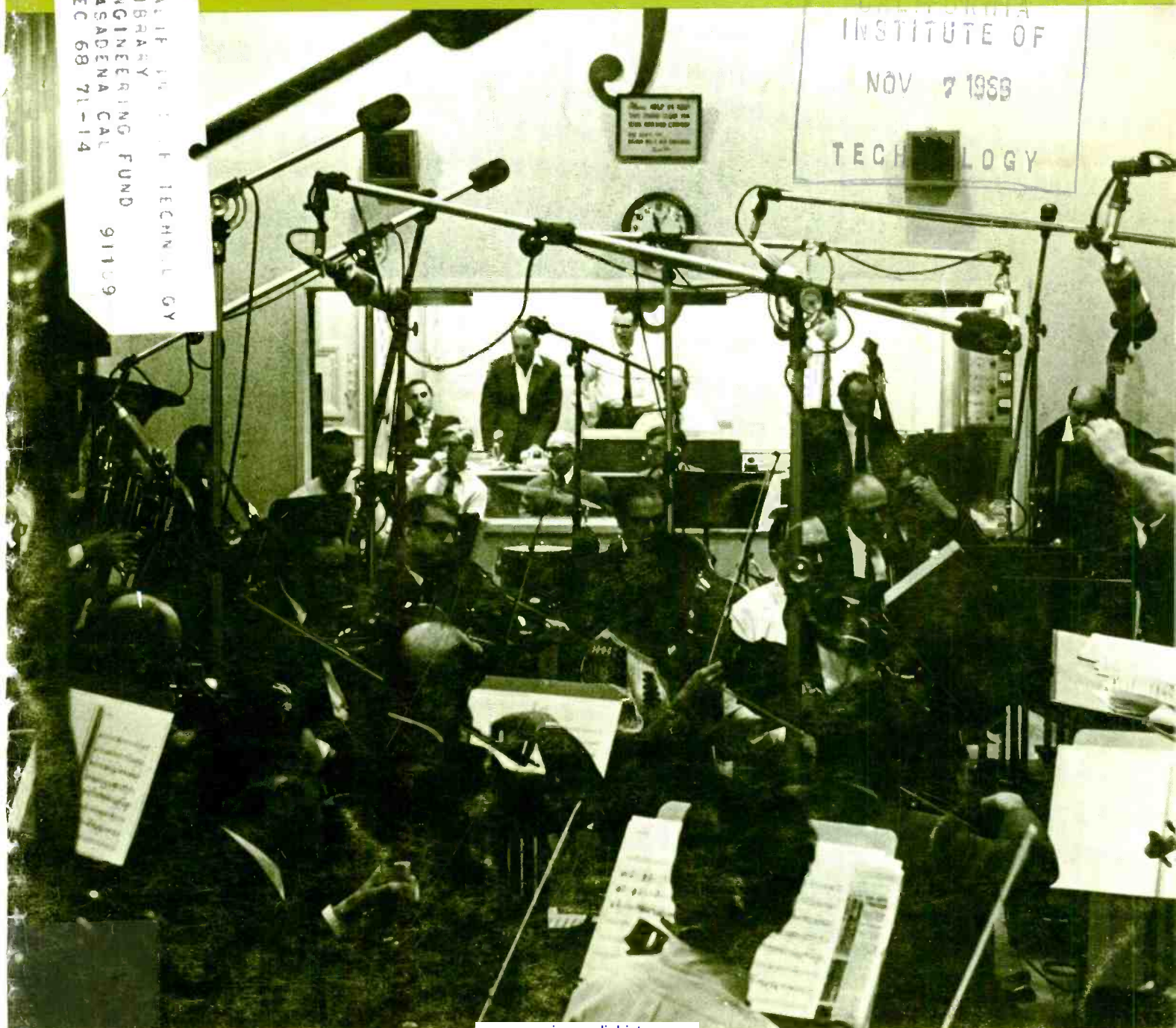
the authoritative magazine about high fidelity

AUDIO

NOVEMBER
1968 60¢

STEREO EQUIPMENT & RECORD REVIEWS

THE WORLD OF PROFESSIONAL AUDIO: Inside a Major Recording Studio ■ A New Method of "Room Acoustic Tuning" The Art of Small Recording Studios ■ Audio Techniques in a Broadcast Station ■ Plus Regular Features



KSJR / KSJN STEREO FM RADIO SAINT JOHN'S UNIVERSITY COLLEGEVILLE, MINNESOTA 56321

DIRECTOR OF
BROADCASTING

February 12, 1968

Mr. Hermon H. Scott, President
H. H. Scott, Inc.
111 Powdermill Road
Maynard, Massachusetts 01754

Dear Mr. Scott:

We have had some interesting experiences with Scott receivers that I thought might interest you. KSJR-FM is a 150,000 watt stereo station broadcasting from St. John's University. Seventy-five miles to the south we operate a second station, KSJN-FM, which broadcasts throughout Minneapolis and St. Paul. This is a satellite station and as such it receives its programming "off-the-air" from KSJR-FM.

In building KSJN-FM we installed a professional rebroadcast receiver. It soon became apparent that the receiver was improperly aligned and that it had several other technical problems. These problems became so severe that we had to take it out of service and return it to the factory. With no auxiliary receiver available, I suggested to our engineer that we might try using the Scott 344 receiver located as a monitor in my office. He reluctantly agreed and we installed the 344 on Thanksgiving. Since that time it has operated in an unheated metal building in its walnut cabinet in weather as cold as 25 below zero, twenty-four hours a day. We feed our broadcast lines directly from it and we have not had to tune the unit more than once or twice since it was installed.

This past week we conducted a survey of our listeners in Minneapolis and St. Paul and I will list some of their comments:

> "The quality of your signal is superb and so are your musical programs"; "The biggest problem at the beginning of your operation was the poor quality signal. With the solution of the technical problems, you have undoubtedly the best radio station going"; "The quality of sound emanating from your station is especially good"; "The sound here in Minneapolis is especially good".

I think comments such as the above are particularly interesting in view of the fact that all of Minneapolis and St. Paul are served by the signal from one Scott receiver.

Sincerely,

William H. Kling

William H. Kling
Director of Broadcasting

November 1968 Vol. 52, No. 11

AUDIO

Successor to RADIO, Est. 1917

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Coming in December

The Commonality of Speaker Systems and Musical Instruments—Antony Doschek discusses the characteristics of wind instruments as compared to horn speakers.

Twenty (Hi-Fi) Questions—Leonard Feldman answers the twenty questions most frequently asked by attendees at his annual Novice Seminar, presented by him for the past five years at the New York High Fidelity Music Shows.

Build a Presence Control—Here are construction plans for a simple mid-frequency tone control that would enable one to "move" a vocalist on a recording into the foreground.

Altec Acousta-Voicing, Part II—Don Davis concludes his description of a modern method of "tuning" a room's acoustic properties with a practical discussion of how it's actually done.

. . . and more

EQUIPMENT PROFILES:

Sony Model STR-6060 AM/FM Stereo Receiver

Sony Model TA-2000 Stereo Preamplifier

TEAC Model A-4010S Stereo Tape Deck

. . . plus others

ALSO:

Record and Tape Reviews, ABZs of FM, Audioclinic, Tape Guide, and other regular departments.

ABOUT THE FRONT COVER:
A forest of microphones partially masks musicians during a recording session at Fine Recording, New York City. See candid comments of Bob Fine on professional recording techniques, starting on page 8.

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.

Eliminating Skating

Q. My tonearm has no anti-skate device. My problem is that it is impossible to adjust the tonearm so that it does not pull slightly towards the center of the disc. How can I correct this?—John Donmoyer, Allentown, Pa.

A. If you can adjust your tonearm so that the inward pull toward the center of the disc is only very slight, you probably have done all that is needed. You are luckier than a lot of us. Some arms cannot be adjusted so that the amount of inward pull is only slight. When the pull is severe, the distortion, especially on the right channel, is increased considerably over that of the left channel. However, if the inward force is only very slight, the amount of distortion on the right channel will not increase sufficiently to cause you to hear it or to create excessive record wear.

One scheme to overcome this problem is as follows: It involves a glass rod mounted just above, and to the right of the tonearm, near the pivot end. Just forward of the pivot, tie a piece of thread on the arm. This thread will come up over the rod and hang down. The free end of the thread is tied to a weight whose amount will depend upon the amount of skating force to be overcome and, of course, upon the angle made by the rod and the tonearm. As the arm moves, the angle made by the rod and thread changes, thereby changing the amount of neutralizing force provided by the weight. This is exactly what we want to happen, because as the tone arm comes nearer and nearer to the center of the disc, the less neutralizing force is required.

If the thread is tied to the tonearm properly, there will be little effect upon the tracking force with distance of the arm from the center of the disc.

You will need to adjust the amount of weight used and the distance of the rod from the arm so that you can obtain the proper amount of counterforce needed as the arm traverses various parts of the disc.

You can't use an ordinary disc to determine the amount and direction of the arm's motion. You will need a smoother surface on which the stylus can ride and be free to move. Further, the surface must be of such a nature that the stylus won't be damaged. I suggest that you obtain a blank disc of the type that recording studios use. Such discs are mirror smooth.

The solution is not elegant, but it is effective.

I believe that similar schemes are used on commercial tonearms. However, the arrangement was taken from a model developed by Mr. Robert Speiden of Rahway, N. J. He is an active and skilled tape and disc recordist, well known for his design of stereo cutting heads.

FM Signal Strength Measurements

Q. For a small proportion of FM listeners, there are instances where one FM station provides all the music we care to hear. I am in that category.

As a long-time reader of AUDIO, *I have read articles on FM antennas, front-to-back ratios, antenna boosters, antenna rotators, signal-to-noise ratio, etc. All of these articles have given me insight into the problems of receiving an FM signal.*

I believe one area has been constantly overlooked, viz., given a good antenna with a fine tuner, how can one measure the signal strength in microvolts of the incoming desired signal, using some simple but sensitive measuring device? It would seem that being able to measure the signal accurately could at least serve as an excellent means for orienting the antenna, certainly far better than the so-called "magic eye" tube. It would be desirable to know what the signal strength is before condemning the tuner to needless investigation. Given the best orientation of the antenna, an adequate knowledge of the signal strength would also provide, among other things, the appropriateness of the receiving location for a particular transmitted signal.—Name Withheld.

A. I would first say that knowing the actual strength of a received signal is not so important to antenna orientation as having a good indicator of relative signal strength. The input to the first limiter is a good place to obtain such information. Fairly sensitive indicator circuits can be easily designed for this application. Various indicator types are possible, but this is a subject for an entirely different discussion.

If your record player today still has a heavy turntable, it must have yesterday's motor

Why did Garrard switch from heavy turntables (which Garrard pioneered on automatics) to the scientifically correct low mass turntable featured on the SL 95? Simply because the synchronous Garrard Synchro-Lab™ Motor has eliminated the need for heavy turntables, which were developed to compensate (by imparting flywheel action) for the speed fluctuations inherent in induction motors. The light aluminum turntable on the SL 95, precision matched to the kinetic energy of the Synchro-Lab Motor, effectively relieves weight on the center bearing and reduces wear and rumble in this most critical area. And its full 11½" diameter gives your records proper edge support.

The Synchro-Lab Motor has also made variable speed controls as obsolete as they are burdensome to use. The synchronous section of the motor eliminates the fluctuations in record rotation which cause music to drift on and off key. It guarantees completely constant, unwavering speed regardless of voltage, warm up, record load and other variables. By *locking in* to the fixed, rigidly controlled 60 cycle current (rather than varying voltage), the synchronous motor insures unwavering musical pitch. And this brilliant new Garrard motor also incorporates an induction section that provides instant starting, high driving torque and notable freedom from rumble.

Garrard innovations such as the Synchro-Lab Motor and new turntable are characteristic of the achievements that make the SL 95, at $129.50, the most advanced record playing unit available today.

For a Comparator Guide, describing all Garrard models, write Garrard, Dept. H2-8A, Westbury, N.Y. 11590

Garrard®
World's Finest

Of course, if you are attempting a field study of a given location in terms of its suitability for the reception of a given station, you would, indeed, require a calibrated source. That problem is not a simple one because you would need some kind of standard signal strength upon which to calibrate your equipment.

Probably the easiest thing to do is to obtain a calibrated signal generator. Connect a meter to the first limiter grid of your tuner, or to the input circuit of the first limiter if it is a solid-state unit, and calibrate this meter in terms of signal strength received at the antenna terminals of the tuner with a dipole antenna connected. The signal generator would have to be made to radiate some signal into the antenna. You still won't have an absolute standard, though you will have an idea of the linearity of your tuner's operation, and you'll have some idea as to whether or not a signal is weak or strong.

I think that considerably more accuracy will be obtained, and with less likelihood of unwanted interference to other nearby receivers, if you coupled the generator right into the antenna terminals loaded with its proper impedance, and the antenna disconnected. Now you can directly calibrate the meter already described in terms of input signal strength.

This method will give you some idea of relative signal strength and some idea of the actual amount of signal reaching the antenna terminals. Of course, field measurements are often made in terms of microvolts per meter of antenna length and this would be quite difficult for you to work out so far as I can see right now.

One word should be said about the meter connected to the first limiter. Whatever circuit is employed to drive the meter, it should not load down the limiter. Improper operation of the tuner will result. It is for the same reason we have high resistance in our VTVM input circuits. You might find it convenient to have several ranges for this meter, similar to the operation of a VTVM. I think you can improve the accuracy of your readings this way. Further, you can have a meter which deflects further on any given range than it would otherwise do. This will enable you to see changes in signal strength more easily.

The arrangement we have been discussing does not quite fulfill all of your requirements. You had hoped that this piece of equipment would indicate the strength of the received signal so you would know that tuner is working right or not. If you make your tuner double as a signal measuring device and if the tuner breaks down, you have lost both the tuner and the "field strength indicator." When the tuner is repaired, you won't know if it is calibrated the way it was originally.

The strength of commercial stations is fairly constant. Therefore, if you are at your receiving location and there is a degradation of the strength of the received signal, the trouble is more likely than not to be generated within your tuner and not in the FM transmitter.

When you come down to it, if I were to explore a potential receiving site, I would use the tuner I expect to use at that location. If it does not receive the station in which I am interested to my satisfaction, it really does not matter very much what signal strength you have available. This is your tuner and it is the instrument you will use.

Of course, if you knew the exact signal strength at your proposed site, and you knew your present tuner was not adequate, you could select one which would more likely do the chore properly.

When I boil it all down, I would also like to have a calibrated unit for the purposes you have outlined, but it is just too hard to make such a device and know that it is as accurate as we would both like. Therefore, I have discussed this matter from several angles.

Audio Techniques

Reducing High Power-Line Voltage

The line voltage here is, to my way of thinking, much too high (128 V.) Hence, motors and transformers run too hot.

To alleviate this condition, I used an old power transformer about 4 by 5 inches—first removing the old windings—to wind 60 turns of No. 12 Formvar to make a swinging choke. I put this choke in series with the line. This gadget cut down the voltage from 128 V. to 115 V.

Now all motors and transformers run cooler than before — *George R. Kirk, Stockton, Ill.*

Reducing Hum

Many writers-in, it seems, have severe hum problems.

Hum is no problem if a person attacks it in a straightforward manner, using logic and not minding having to run a stout lead *each* from turntable, tape deck, tape recorder, tuner, amplifier, and preamplifier to a single, solid ground which is not shared by any other appliance unless it is a direct part of the particular system.

I bought an eight-foot long, copper-clad steel rod, drove it into the ground, and used some rocks until only a small nubbin for the bronze clamp remained above ground. Any gear which had a three-wire grounded plug I lifted off ground by using an adaptor socket with the green wire and spade lug cut off short. From *each* chassis I took out to my ground stake a piece of no. 14 stranded wire for its own *separate* ground lead.

These direct, simple, deliberate steps have reduced hum to a state where the tubes' "shot noise" is a mighty roar by comparison. The ground is also cheap by comparison with total system cost. —*Will Cochran, Hatfield, Pa.*

Mounting Large Electrolytic Capacitors

Editor's Note: This submission came as a result of a discussion concerning mounting of two 4500-μF 50-V electrolytic capacitors ("Audioclinic," Feb. 1968).

In my experience, the use of fibre or rubber grommets to mount large electrolytic capacitors by their screws is a risky and sloppy technique. A better technique to mount these 4500-μF capacitors is to cut approximately 2⅛" holes in the chassis, through which the 2$\frac{1}{64}$"—diameter, 4$\frac{9}{64}$"—long capacitors are inserted. The capacitors are then affixed by the use of proper mounting clamps (that is, Mallory VR8 clamps). — *George E. Mayer, Troy. N. Y.*

Restoring Old Recording Blanks

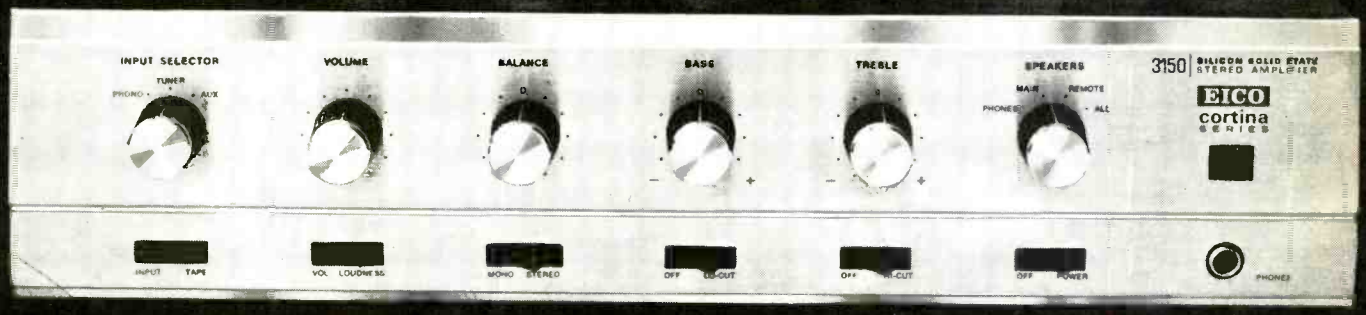
I had a package of old Presto discs that would not cut at all, even with a hot stylus. I was able to salvage them to a remarkable degree. I dipped them in an acrylic retarder. The result was that they were too soft to use for a few days or weeks (I forget which) but seemed to settle down and, while not as good as new discs, work very well.

Also, by this same process I have been able to re-use used discs (recorded microgroove), as the fine grooves run together and a fairly good surface results.

Unfortunately, I did not complete my experiments, but just went far enough to be encouraged. My process was crude. I am waiting for the time when I can dip them in a more dust-free environment.—*John M. Kaar, Menlo Park, California*

This is more amplifier than you may think you need.

But after you see the price, why settle for less.



The EICO "Cortina 3150" all-silicon solid-state 150 watt stereo amplifier is truly a lot of amplifier. It combines wide-range preamplifiers, controls, and power amplifiers, all on one uniquely compact chassis. It delivers clean power to *two* sets of speaker systems, stereo headphones (for which there is a jack on the front panel) and a tape recorder. The Cortina "3150" gives you *complete* control facilities.

Most people think that, while all this would be very nice to have they don't want to pay a lot of extra money for it.

We agree. That's why we designed the "3150." Fully wired it costs $225.00. If you want to buy it as a kit — and it is a particularly easy kit to assemble because of our advanced modular circuitry techniques — it's a mere $149.95. The beautiful Danish walnut vinyl clad cabinet is included at no additional cost. At these prices, the "3150" is no longer a luxury. It's virtually a necessity. The power delivered by the "3150" is enough to give faithful reproduction of the highest peaks in music even when it is used with inefficient speaker systems.

The "3150" gives you more than just power. With *both* channels driven the harmonic distortion is *less* than 0.1%, IM distortion is less than 0.6%, frequency response is ±1.5db, 5Hz to 30 KHz, all at full output, hum and noise 75db below rated output; channel separation is more than 50db; input sensitivity is 4.7MV at magnetic phono input, 280MV at all other inputs.

Phase shift distortion is negligible due to the differential amplifier input circuit and the transformerless driver and output circuits. All electronic protection (no fuses) of output transistors and speakers makes overloads and shorts impossible.

The "3150" also provides ten versatile control facilities: volume, balance, full range bass and treble controls. Input Selector (phono, tuner, aux), tape monitor, loudness contour, low and high cut filters, and speakers system selector switches.

See and hear this most advanced of all silicon solid-state amplifiers at your EICO dealer. We are confident it will quickly change your mind as to how much amplifier you really need.

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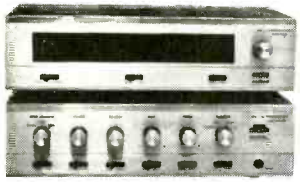
Designed, manufactured in U.S.A. and guaranteed by EICO.

See the complete Cortina® Line at your EICO Dealer.



"The EICO Cortina Series are low-cost audio components that look and sound like high cost components."
Popular Science Magazine.

Cortina 3070 a full capability 70-Watt All Silicon Solid-State Stereo Amplifier for $99.95 kit, $139.95 wired, including cabinet.

Cortina 3200 Solid-State Automatic FM Stereo Tuner for $99.95 kit, $139.95 wired, including cabinet.

Sound n' Color is an exciting innovation in the home entertainment field. It adds a new dimension to your musical enjoyment. See the music you hear spring to life as a vibrant, ever shifting interplay of colors.
See every tone, every note, every combination of instruments create its own vivid pattern.
EICO Model 3440—Kit $49.95, Wired $79.95.

Cortina 3570 70-Watt All Solid-State FM Stereo Receiver for $169.95 kit, $239.95 wired, including cabinet.
Pre-assembled and pre-aligned front end, multiplex, and IF circuits speed and simplify assembly — you'll enjoy building this deluxe kit.

New Cortina 3770 70-Watt All Solid-State AM/FM Stereo Receiver for $189.95 kit, $279.95 wired, including cabinet.

EICO Electronic Instrument Co., Inc. A-11
283 Malta Street, Brooklyn, N.Y. 11207
☐ Tell me where I can have a free Cortina demonstration.
☐ Send me free 32-page catalog on EICO's 200 "best-buys."
Name_____
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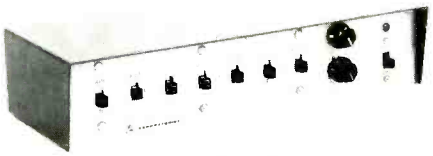
What's New In Audio

C/M Electronic Crossover

C/M Laboratories debuts its Model 601 electronic crossover, designed to give the user any crossover frequency from 100 Hz to 12.7 kHz in 100-Hz increments. A level control for relative adjustment between high- and low-pass amplifiers equalizes efficiencies of loudspeakers. The system is said to reduce crossmodulation distortion between high and low frequencies in a single

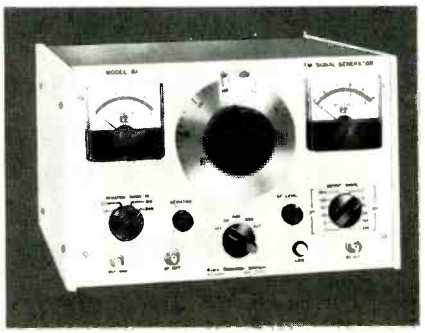


amplifier. Two Model 601s can be cascaded for either steeper crossover slopes (crossover point of a Model 601 exhibits a 6 dB/octave slope) or for tri-amplification purposes. The solid-state unit has 4 MOSFETS, 8 other transistors and 4 silicon diodes. Total harmonic distortion is 0.5% at rated output (2 volts rms). Dimensions are 2½" H x 11⅛" W x 5¼" D. Weight is 4½ lbs. Price, $126.00.

Check No. 8 on Reader Service Card

High-Quality FM Generator

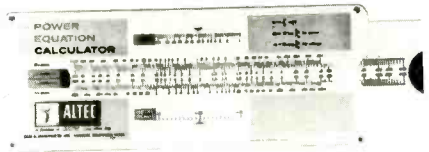
A new variable-frequency FM generator has just been introduced by Radio Research Company of Rockaway, N. J. Long a supplier of fixed-frequency units for factory use, the company now has employed most of the features of their reliable line of generators to the Model 61, which covers the range from 85 to 135 MHz, thus covering the FM band and the possible image-frequency range of conventional FM receivers. The unit provides an accurate output from a full-scale indication of 1 μV to 0.3 V, and four ranges of deviation with full-scale indications of 10, 25, 100, and 250 kHz. Two internal modulation frequencies are provided—400 and 1000 Hz—and external modulation is acceptable to accommodate stereo generators. Leakage is practically unmeasurable, and modulation fidelity is flat within 1 dB from 30 Hz to 100 kHz. Solid state throughout, with minimal "warm-up" time and high thermal stability, the unit is suitable for production line use where constant changing of output level does not wear out the instrument and change indications. Price, $695.00.



Check No. 10 on Reader Service Card

Power Equation Calculator

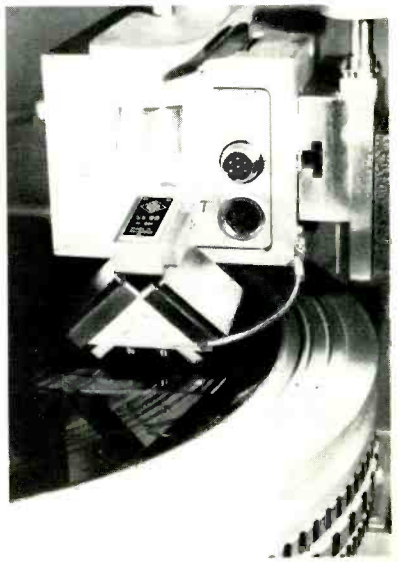
An 8½" long, double-sided Power Equation Calculator, designed to solve



problems in power and decibel equations, is available from Altec Lansing, 1515 So. Manchester Ave., Anaheim, Calif., for $1.00.

New Neumann SX-68 Stereo Cutterhead

The new SX-68 Neumann stereo cutterhead features (1) secondary resonance that is above 75 kHz, (2) high feedback capability for a frequency-response deviation less than ±1 dB, as well as extremely low distortion over the entire audio range, (3) channel separation better than 35 dB over the full range, vertical tracking angle of 15 deg., and (4) non-critical stylus change. The SX-68 is also fitted for Helium cooling, with which maximum lateral velocities at 10 kHz is 40 cm/sec (2.7 amperes) for 10 sec or 26.5 cm/sec (1.8 amperes) continuous sine wave.



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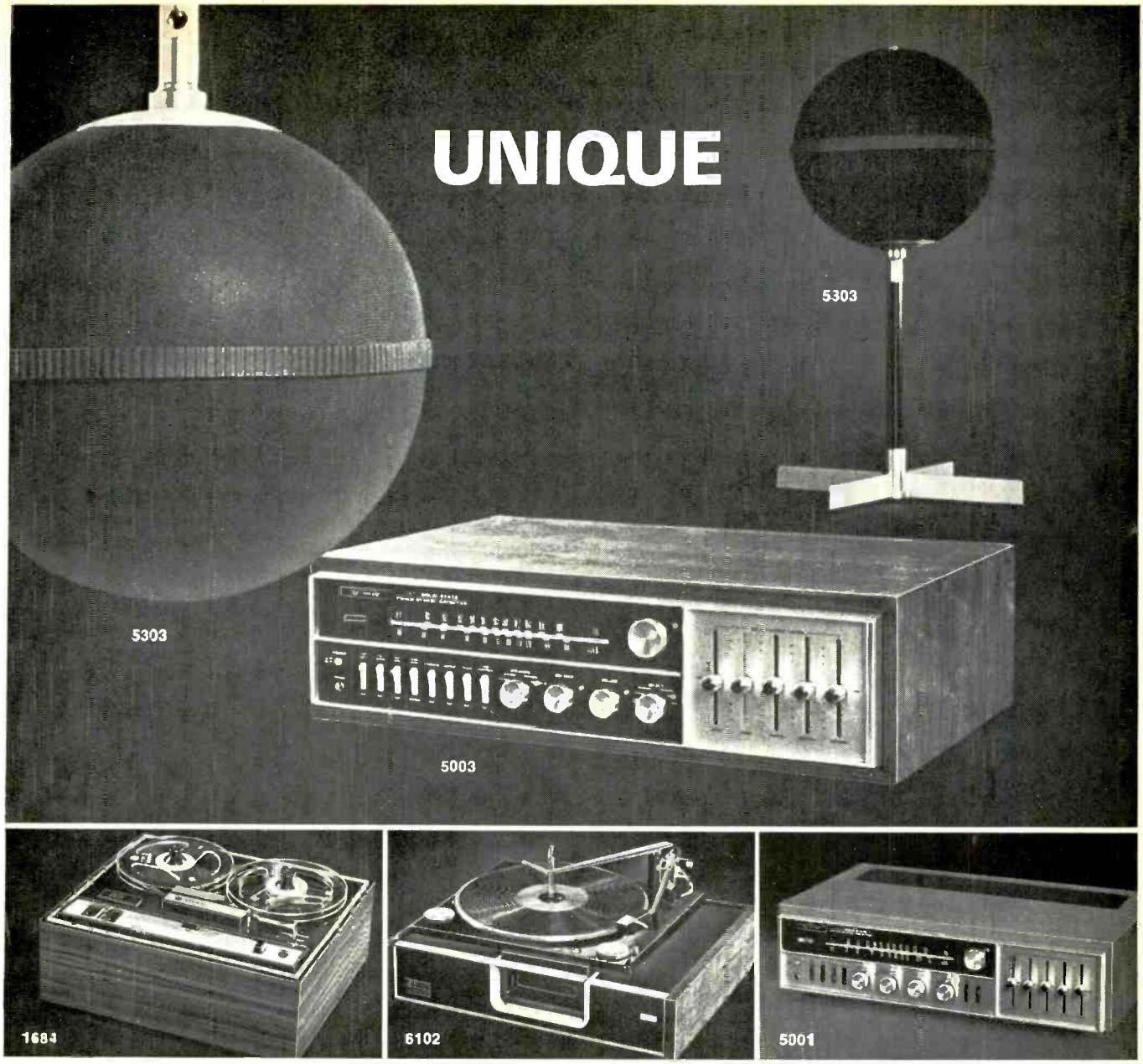


Shown above and described below are just a few examples of the most unique and formidable line of stereo equipment in the world today. From powerful stereo systems, to all-in-one compacts, to breathtaking individual components, there is a model designed for everyone from the most ardent stereo enthusiast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

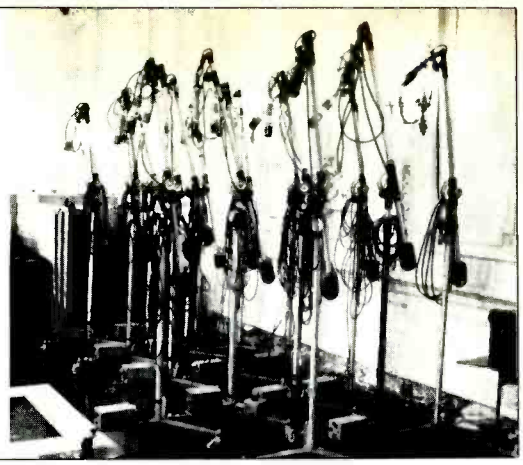
For additional information and a copy of our new 28 pg. full color catalog, write: Dept. J,

JVC America, Inc., A Subsidiary of Victor Company of Japan, Ltd., c/o Delmonico International Corp., 50-35 56th Rd., Maspeth, N.Y. 11378, Subsidiary of TST Industries, Inc.

BEHIND THE SCENES

BERT WHYTE

The Professional Viewpoint



A dialogue with Bob Fine, colorful and controversial head of Fine Recording, New York City

In the heart of the "silk-stocking" district of New York City, across the street from proud Steinway Hall and surrounded by the chic boutiques and swank shops that cater to the "beautiful people," stands the Great Northern Hotel. It is old and begrimed, and gives off an aura of genteel shabbiness. In this unlikely locale, the modern studios of Fine Recording occupy some 38,000 sq. ft. of space. A special elevator took me directly to the reception area on the penthouse floor, which is part of the Fine Recording complex, and shortly thereafter I was escorted into the panelled office of Bob Fine, president of Fine Recording. Here the results of our talk:

Bob, you've been in the recording business since you were a boy of fourteen, am I right?

FINE: Yes, in 1938 I worked for Miller Film in Steinway Hall as a general flunky and as a "wax shaver." In those days we were still doing direct mastering on cakes of wax, and my job was to prepare them for recording.

Is that the same Miller whose cutter you used years later when you were recording the Mercury "Olympian" classical recordings?

FINE: That's right. You may remember how appalled recording people were because Miller wanted a thousand dollars for the cutter. What a change from these days, when a Westrex stereo cutter runs more than four thousand dollars!

I believe your first studio venture was in 1951, after you had left Reeves Sound.

FINE: Right. I was quite happy at Reeves, but I always had a yen to have my own studio and thus Fine Sound came into being in Stony Point, New York. Most of the equipment was in a converted chicken house and the disc-cutting operation was in a former barn. A lot of the recording people resented having to make the trek "up the mountain," but we prospered after a fashion.

It was while you were located at Stony Point that you contracted to do the new Mercury classical series, which gave birth to your recording truck. Wasn't it about this time that you invented Perspectasound?

FINE: Well, you may recall that the movie industry had been taking an awful licking from television, and they were desperately searching for ideas and gimmicks to stem the TV tide. Thus we had the rash of three-dimensional movies and finally *Cinemascope*, which with the wide-screen aspect ratio, virtually demanded stereophonic sound. The trouble was that true stereo was almost impossible to incorporate into a movie because of the constantly changing perspective. In the long run, it really came down to was that the most practical thing was an emphasis on lateral directionality because of the wide screen. I figured the way to do this was by post mixing after the picture was finished. By means of panning pots and pilot signals to activate left, center and right loudspeakers behind the Cinemascope screen, with deliberate "built-in" directional sound, we could lick the perspective problems. *Perspectasound*, then, was directional sound on an optical track, which was compatible with standard film reproduction, so all that was necessary was to make one negative. In theatres equipped with the *Perspectasound* circuitry and three speakers, a quasi-stereo effect was obtained.

I take it that this is when Loew's/MGM showed interest in Perspectasound?

FINE: Actually we had shown the idea to several film companies, but after a demonstration of the system in an abandoned movie house in the Bronx, MGM acquired *Perspectasound* and Fine Sound and set up our operation in elaborate studios at 711 Fifth Avenue in New York.

A Fifth Avenue studio was a pretty far cry from a chicken house in Stony Point. One thing I remember vividly about that studio at 711 was the fabulous aromas that used to waft up the elevator shaft, since the kitchen of the famous "Le Pavillon" restaurant was on the ground floor. If I remember, you did some pretty big pictures in the Perspectasound **process at 711.**

FINE: I guess the best of them was the Marlon Brando "Julius Caesar."

I understand that you became emeshed in some sort of patent fight and unfortunate legal problems, which resulted in the ultimate demise of Fine Studio.

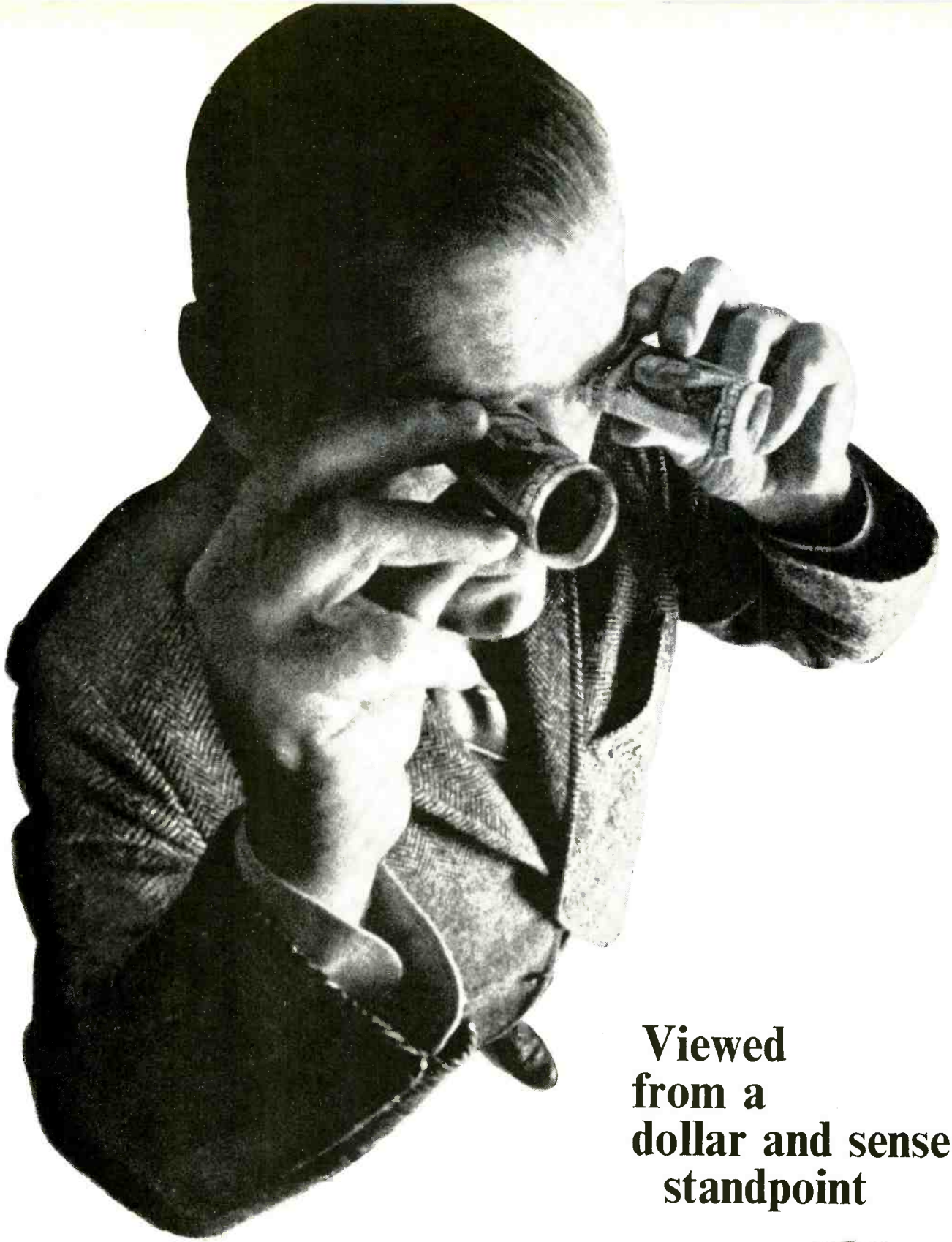
FINE: All too true. I never received a nickel on my patents, even though *Perspectasound* is still used in some foreign theatres. In any case I wanted to start another studio, so to raise money I went to Europe and designed studios. This was just about the beginning of stereophonic recording in Europe in 1956, and I did two in Paris, one on Milan, and several others. I had my lawyer looking for possible studio locations in New York, with the proviso that it should have a ballroom. Shortly afterwards he came up with the ballroom of the Great Northern Hotel; we signed a lease giving us exclusive use of the ballroom and, thus, Fine Recording was born.

Speaking of leases, how secure is your present lease in view of the fact that you are in an old building, with the present tendency to tear down the old and build still another glass and aluminum skyscraper?

FINE: We have leases running to 1980, and the plot of land the hotel occupies is not in itself big enough to build on. Someone would have to put together a large expensive parcel of land, in addition to the hotel plot, to make a new building a practical proposition.

I assume this is why you are going ahead with your present expansion plans?

FINE: Yes, we now have 38,000 sq. ft. in this building. We occupy the total top of the hotel where we have two mixing theatres, all the disc cutting operations, production offices, and main reception area. On the twelfth floor we occupy 50 per cent of the space, consisting of an office wing, another studio wing, the automated mastering section, the electronic music operation (the Moog synthesizer is here) and a research wing where we carry on research in



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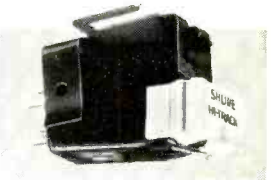

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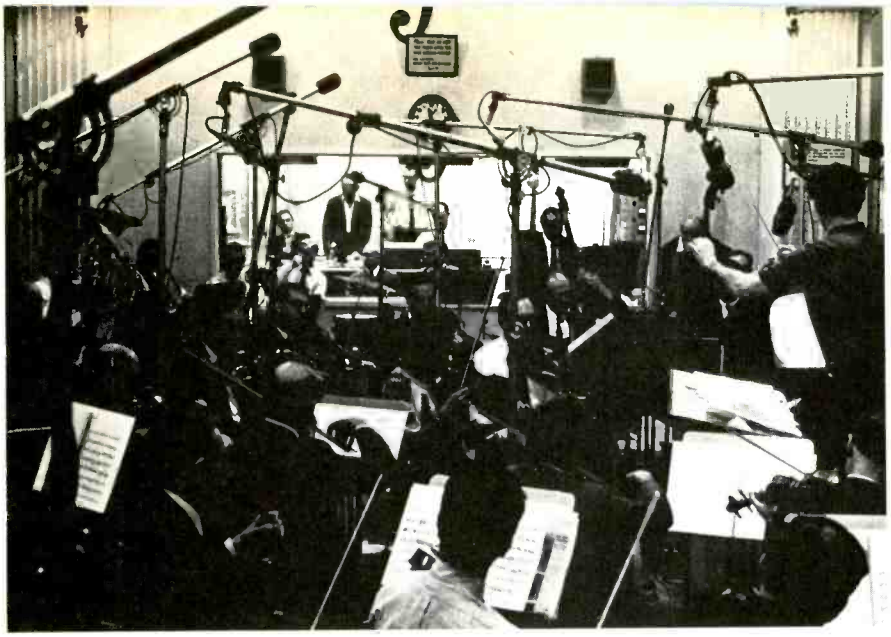


Fig. 1—Recording sessions in Fine Recording's Studio "A"—the ballroom.

audio and in other electronic but unrelated areas. The lobby of the hotel is where the main studios are located. And our tape duplication plant is in the basement area. When we took over the ballroom, which is now our Studio A, it was a dirty mess, painted a garish red. It had an old oaken stage where the N.Y. Philharmonic used to rehearse. The chandeliers were broken, the wooden panelling largely destroyed. Two years later we restored the ballroom to its former glory, replete with virginal white paint and gilt, as well as new chandeliers.

What changes did you make to turn it into a recording studio? I understand there were large columns in the room which some people criticized for breaking up working space.

FINE: We have had some complaints. Nevertheless we have been able to accommodate 60- to 70-man orchestras for symphonic recordings and motion picture scores with a minimum of difficulties. We built a control room endeavoring not to change the acoustics of the room. The huge old oaken stage was removed and replaced with a smaller stage which we use when it is necessary to elevate an orchestra. As far as I am concerned, the acoustics make up for the small inconveniences.

What is the reverberation time of the ballroom?

FINE: That is an interesting point. Measured by the Academy standard reverb scale, the reverb time is about three seconds.

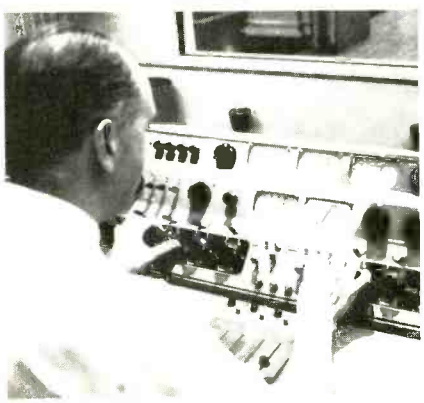
Isn't that fairly long for recordings?

FINE: Yes, but I don't believe in that reverberation measurement. In the number of times I've spoken to Leo Beranek, and with all due respect to that learned gentleman and his accomplishments, I've disagreed with him on this standard of reverb measurement which is calculated on the basis of how long it takes from the initiation of a sound until it measures 60 dB down. I think this is quite wrong . . . you do not sense reverb past the point where the sound is down 15 dB. You just don't sense it, or hear it, or feel it.

Then you feel that according to your system of measurement the ballroom has suitable acoustics. Measurements aside, don't you feel that the real burden of proof is how your recordings actually sound?

FINE: That's true, but just for the record I think we have slightly over a second of usable reverb. Of course there is still the matter of the frequency spectrum in the room. At what frequencies does the room reverberate? Some rooms have huge peaks at say 400 cycles [sic] or standing waves which by oscillatory reinforcement can make a room appear to have more reverb. I am considering that you must take the frequency spectrum into the reverb picture and measure from about 800 cycles upward. Below 800 cycles, reverb just isn't significant.

Fig. 2—Chief Engineer Ted Gosman at controls of the console used for Studio "A" sessions.



In addition to the ballroom which is your Studio A, I notice a Studio B just off the hotel lobby. Would you describe it?

FINE: Studio B used to be the serving kitchen for the ballroom. We cut it off from the ballroom and put in an isolation wall which weighs several tons, but which is in a "floating" mode to avoid sound transmission. Studio B is fairly small, but with a high ceiling. We use it a lot for voice recording and for rock-and-roll dates and orchestras. We think it has a good sound, but it is a contrived studio, whereas Studio A has its own completely natural sound.

Getting back to the ballroom, when you do "pop" dates there, do you use the natural reverb or do you augment with reverb chambers?

FINE: We use EMT's reverb chambers because, with the sound preferred in today's pop market, with ears attuned for exaggerated reverb, it is too difficult to use natural reverb. Then, too, if you have a singer in a live room, close to a microphone . . . where you want isolation . . . you eliminate so much of the natural reverb you must make up for it with EMT's. Unfortunately, when you are recording rock and roll and all these wild, screaming, brassy things . . . a live studio is extremely difficult to handle. For rock and roll the most successful approach seems to be these low-ceilinged, very-dead studios, where they don't really do conventional mixing and can control the sound quite readily. I like the sound of a big wild orchestra in a live room, but it is difficult to mix and achieve instrumental isolation. However, when you talk of isolation in stereophony, the acoustical coupling is important and many people really don't like that.

I understand you have recently installed 8-track recorders in the ballroom and that this solves acoustic problems to a certain extent.

FINE: Since we have installed the 8-track Ampex and will soon be installing 12-track Scully recorders, the ballroom has sort of been "rejuvenated" for pop recording. You see, we no longer record in the classic sense of the word. Now we lay down against an electronic click track [A click track is like an electronic metronome with beats down to a 32nd note, but 1/8ths notes are usually used] . . . the rhythm on say two or four tracks . . . the drums, bass and guitar, and any other rhythm instrument, and we record a basic rhythm track on the machine. Everyone works with earphones . . . no one

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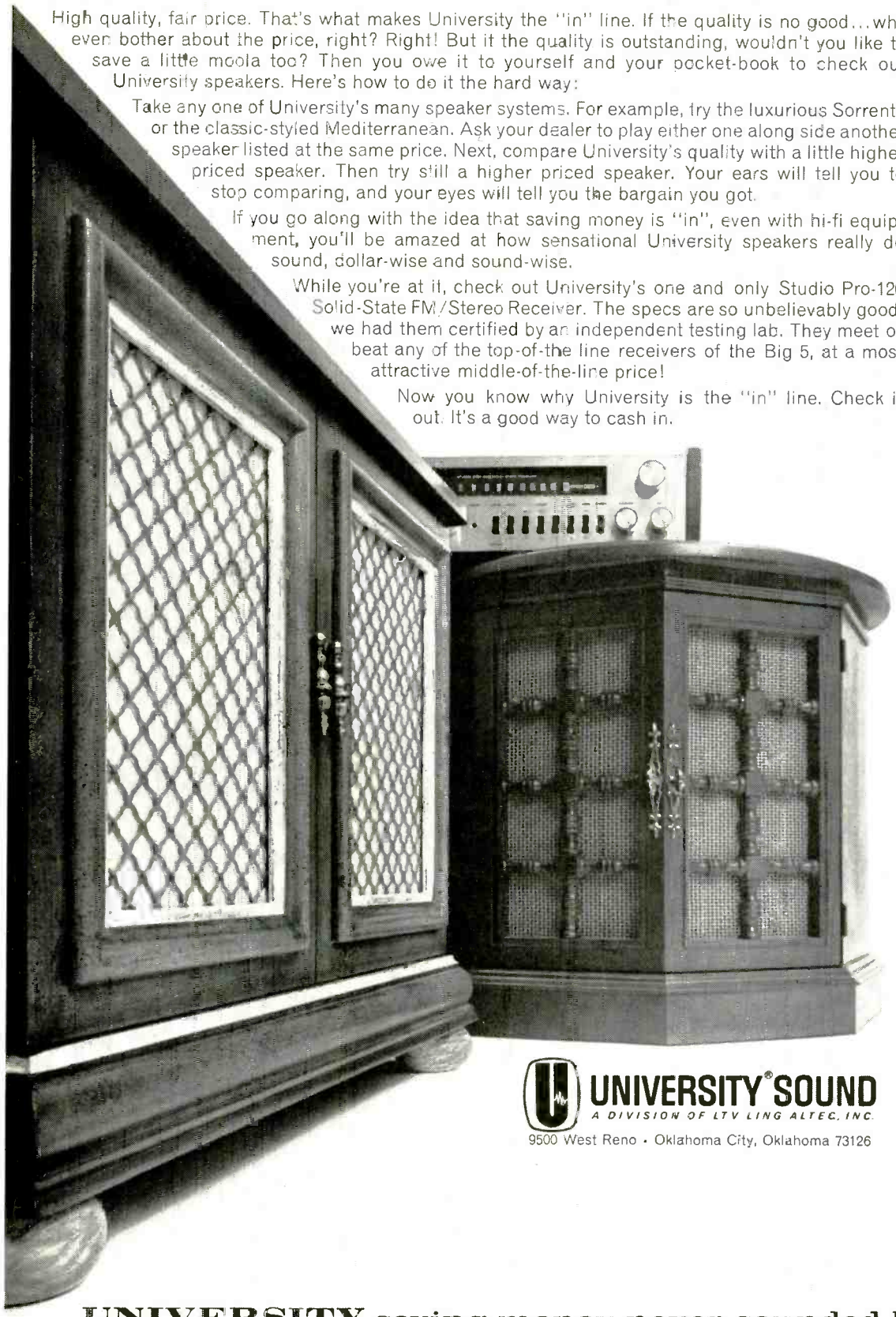

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works live anymore . . . everything is recorded against these electronic metronomes. We may put out 30 to 40 earphones on a big date . . . one for each musician. As I said, we make a rhythm recording on 2 to 4 tracks of the eight-track machine, put the machine into "sel-sync" operation [sel-sync (selective synchronization) works this way: what has previously been recorded is played back on the record head . . . the record head becomes a playback head . . . and a 4th head records the new material or new track in synchronism with the previously recorded sound. It is really just an elaborate head switching system. After each track is recorded, you rewind to the start and you are ready to add another track or tracks, as the case may be. As you can see, the possibilities are limited only by the number of tracks available.] If there is a woodwind section we sel-sync record them on another track or tracks, as required. The woodwind instrumentalists hear the rhythm tracks through their earphones and play against the rhythm. Next we sel-sync the brass on one or more tracks in the same fashion as the woodwinds. If there are voices they are usually recorded sel-sync on tracks 7 and 8. The voices are recorded singing the same material on the two tracks, but at different times. They sing the same harmonies, etc., but when you mix the tracks together, because their timing is not that precise due to the human element, you get this bigger sound which enables voices to compete acoustically against the huge sounds of brass and rhythm.

Bob, with so many tracks being used for the various elements in a recording, I can see now why you are installing 12-track machines and would probably like to put in that monster 24-channel Ampex which uses tape two inches in width, if you could afford it.

FINE: Well, I think it tends to get ridiculous after a certain point. You could put practically every instrument on its own track with the 24-channel unit . . . and in certain of these wonderfully complex rock-and-roll recordings they do just that . . . recording sounds that are humanly impossible to produce on a standard recording basis.

Once all 8 tracks are recorded, what is your next step?

FINE: We go into a re-mix situation where we literally re-create the session from the stored information. That can go on for hours and hours . . . which for a studio is economically very nice . . . it's a new thing which never was done in studios before. It is a new market because, after all, in the recent past you mixed a date and, outside of a few balance adjustments, that was that; there was no further processing. Now you have 8 or 12 separate sources of information and it's a whole new world of recording that can be either abused or used constructively.

I presume that, once you mix all the tracks, it is brought down to a two-channel stereo recording for the ultimate consumer product, be it disc or tape in its various formats.

FINE: You know it all used to be mixed right down to monophonic, but with the demise of mono recordings, we do two-channel stereo, unless it is for a foreign market where mono still is predominant.

At this stage, I suppose the A&R producer is the one who decides where the various tracks should be placed in terms of left/right stereo orientation.

FINE: Yes, and he may also decide to "pan" some tracks . . . all of our consoles are equipped with panning pots . . . he may want an effect like a marching drum. What the producer may desire may be so complex as to require another 8-track mix from the original 8-track recording.

Returning to the ballroom, Bob, I take it that a recording in the classic three-channel stereo "symphonic style," with the traditional instrumental positioning, is quite rare these days?

FINE: Certain people still like the "pure" 3-channel approach, and we do some occasionally. Nowadays most producers are of the type where 50 men can be playing and they want to hear an electric harpsichord isolated, and at a louder level than the group. It is incongruous, but it is typical of today and is at least legitimate in terms of an interesting sound, if nothing else.

What are your feelings about the use of compressors or limiters?

FINE: When you put any limiting equipment that is of a type where you rectify the output of an amplifier and appply it back to change the gain of the input stage of the amplifier, you are seriously affecting pulse information. That is one of the reasons that 8 or 9 years ago we changed our whole compression system. We do not use an RC-type compression system. Our compressions systems are based on the factor that the long-wave information is redundant and is consuming the greatest power. Our compressors pass pulses and act on long-wave information. The result is great presence on a record or a film without having a tremendous amount of electrical energy to deal with. With a normal type of compression system you can hear the "gating." Just jangle a key ring or put maraccas in front of a mike with the compressors on and listen to what you have at the output. In cases of very efficient modern compressors, you don't hear the keys or the maraccas unless they are down below the thresholding levels of these instruments. Put a piano through this kind of compression and you can turn a piano into an organ!

Do you ever use limiting or compression on classical material?

FINE: We do have to in some cases of extreme dynamic range, which we have done for years. We do have to use manual monitoring; this is always done with A&R people . . . not with engineers. There are real-time situations where you cannot possibly involve yourself with the dynamic range you have on the tape . . . it is usually impossible to put it on a record. If you make a record with too wide a dynamic range, people can't listen to it . . . or don't want to be bothered with the constant adjustment of controls. We can record a wider dynamic range than we do on any medium that we make . . . but every time we have fought this battle that people should listen at the right level for the proper dynamics, we lost the battle. We therefore temper the dynamic range to within usable levels below normal room levels. Our original Mercury classical recordings had so much dynamic range we used to get complaints from the field that people would lose whole sections of pianissimo music when they adjusted the fortes to the low listening levels prevalent in most apartments and in many homes.

Are you still interested in film work?

FINE: Most assuredly. We have two theatres on the top floor where we do motion picture mixing. This is in our Studio C.

Do you use 35mm magnetic film in the scoring studios?

FINE: We use both tape and film, and can go from one medium to the other merely by switching.

Are your consoles of solid-state design?

FINE: We are in the process of installing solid-state consoles in all the studios. At the moment, the consoles are tube jobs with the exception of a new solid-state unit in Studio D. The console in the ballroom has a turret that controls reverb. I mention this because we used to have concrete reverb chambers in the basement, which at one time were food storage lockers. They

(Continued on page 100)

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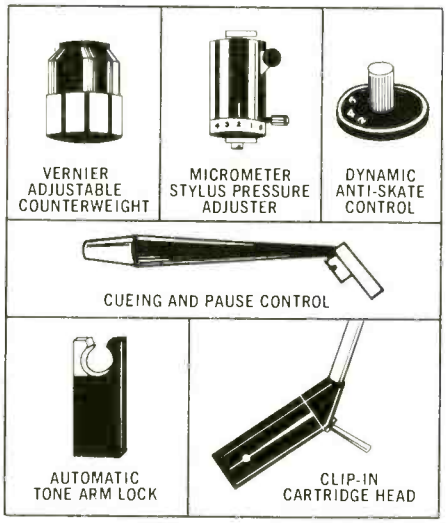

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Tape Guide

HERMAN BURSTEIN

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Going Professional

Q. I have decided to turn my tape recording hobby into a business, but I am confused by what type of equipment I might buy. I am going to use 12 microphones, with cable runs of 25 to 45 feet to the control room. I would like to have separate bass and treble controls for each mike input, and I would like to use a tape recorder with built-in echo at 7.5 and 15 ips. I would welcome your advice.—Jerald G. Boykin, Mailey, No. Carolina.

A. I do not recommend the use of a tape recorder's built-in echo facility for professional or semi-professional purposes. Such "echo," which consists simply of feeding some of the playback signal into the record head, is only a rough approximation of the effect produced by professional echo devices.

Your desire for as many as 12 mike inputs, together with separate bass and treble controls for each, suggests that you may have to go into a custom-built mixer (made for you or by you). A desirable first step is to find out what is commercially available, which you can do by perusing the catalogs of mail order houses and by writing directly to manufacturers of professional mixing equipment. You may also want to look at the articles on mixers that appeared in the October 1962 and October 1963 issues of AUDIO.

Intermittent Noise

Q. I have an intermittent noise in the playback preamp of my tape recorder, sounding like a heavy scratch in a record at relatively regular intervals. It seems to be in both channels and comes and goes. Sometimes it will start when the preamp is first turned on, and later it will disappear; or it may not be present at first and may appear after a half hour or hour's use. I have tried replacing transistors in the playback preamp, but without improvement. I will appreciate any thoughts you may have.—Lewis J. Thomas, Philadelphia, Pa.

A. There may be inadequate decoupling in your power supply, possibly because of a defective filter capacitor, which leads to oscillation that you hear in the form of a periodic "heavy scratch." If replacing the filter capacitor(s) does not help, investigate other components in the power supply.

Tape Comparisons?

Q. Has any one ever compared the quality of various tapes? It is hard to tell which is best for a given application? — Thomas Higgins, Alexandria, Va.

A. To my knowledge, no one has made an authoritative brand-by-brand, type-by-type comparison of tapes, at least for publication. The best approach still seems to be to try several brands and types made by reputable companies and find out empirically which is best suited to your particular tape machine and needs.

Adjustment of Bias

Q. In the instruction manual for my tape deck, regarding adjustment of recording bias, it mentions to record a 1000-Hz tone, adjust the bias control for peak bias, and then back off on the control ½ dB. This puts the tape deck in an under-biased condition. Publications that I have read on the subject state that many professional tape decks are adjusted for peak bias, while others are adjusted for a 2-dB over-bias. They state that under-bias causes distortion, while over-bias cuts down high-frequency response.

The manufacturer of my tape deck claims fantastic treble response at 3.75 ips. I attribute this to under-biasing. Am I correct? When I adjusted my tape deck for peak bias with a 400-Hz tone, I found that the 3.75-ips high-end response dropped from 15 kHz to 10 kHz. At 7.5 ips the response dropped from 19 to 15 kHz. How should bias on my machine be adjusted?—Don Woodruff, San Rafael, Calif.

A. Under-biasing will extend treble response, at the price of increased distortion. To elevate their claims as to treble response, some manufacturers resort to under-biasing.

The method of biasing tends to differ somewhat among manufacturers, depending upon their philosophy of treble response versus distortion, upon the kind of tape they envision to be used with their machines, upon design characteristics of the machines and heads, and upon considerations of signal-to-noise ratio.

To the extent there is a widespread practice, and one that usually produces close-to-optimum results, it is this: at 7.5 ips a signal of 1000 Hz (some manufacturers recommend 500 Hz) is recorded, and bias is adjusted until maximum signal is obtained in playback; bias is then further *increased* until output in playback is ½ dB below maximum output. Such over-biasing tends to put the tape in a region of response-versus-bias that is relatively flat, so that the tape is not too sensitive to moderate variations in bias that may occur as the machine warms up, as line voltage changes, etc. Usually it does not matter too much whether the signal is 500 or 1000 Hz at 7.5 ips; for both frequencies at this speed (i.e. for the wavelengths in question) the curve of output versus bias is very similar in the region of maximum output.

For a machine seeking a desirable balance among good treble response, low distortion, and low noise, response to about 15,000-18,000 Hz at 7.5 ips and to about 10,000-12,000 Hz at 3.75 ips (down no more than about 2 dB) is pretty much in line with the present state of the art. To an extent, the upper frequency limit depends upon the quality of the tape heads, particularly the playback head.

Automatic Gain Control

Q. I am doing tape recordings of weddings. The officiating Priest, Minister, etc., usually has a tremendous voice, and he generally pins the VU meter. The groom and the bride as a rule speak subduedly, and all three stand in a tight circle. Set the tape recorder for Minister, and I don't get the bride and groom; set the tape recorder for the bride and groom, and the Minister pins the VU meter but good. What can I use to electronically equalize the gain?—Andrew R. Winkler, Newark, New Jersey.

A. Some of the inexpensive tape recorders have an automatic-gain-control circuit. If such a machine isn't suitable for your needs, you might obtain a schematic of the machine and copy that part of the circuit involving automatic gain control. Another possible approach is to use a microphone with a "figure 8" response characteristic (a ribbon microphone) and aim it so that the relatively dead side points most nearly at the Minister. Æ

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Letters

SCA Receivers

• I have just read with interest the article by Mr. Leonard Feldman concerning SCA—Private Music Channels on FM Stereo.

I am wondering why he did not advise his readers that use of an SCA Channel by unauthorized parties is a federal offense, and subject to a sizable fine and violates section 605 of Federal Communications Act.

JOSEPH F. HARDS,
Seeburg Music Library, Inc.
Chicago, Ill.

• *Use of an SCA receiver is unauthorized for* commercial *applications, but there are no restrictions for personal, non-commercial uses. This was noted in an article that followed in the next issue of* AUDIO, *October 1968.—Ed.*

• I read your excellent article on SCA in the September 1968 issue of AUDIO.

I had built a [SCA] receiver for use at home but have never been able to completely separate the main channel audio from the SCA music. At normal listening levels it can be barely heard, but it is there.

Recently I reviewed the fundamentals of the transmission spectrum transmitted and wonder if I have hit on the reason. In your diagram of the spectral distribution of information on an FM channel you show the L-R sideband as ending at 53 kHz. However, I am wondering if the maximum deviation of the 53 kHz due to the 75% excursion possible due to amplitude of 15 kHz would send that limit out to 67.5 kHz. If this is the case, the maximum separation possible to achieve, regardless of the selectivity of the circuitry employed, would be limited to less than 40 dB. This would mean that the main channel could be heard in a quiet room with normal listening level of the SCA music.

Am I right or do I misunderstand?

SALVATORE A. ROMANO
Brooklyn, N. Y.

Author's Reply:

• I have run into much the same problem with respect to cross-talk from main channel into sub-channel, but suspect the cause lies in several other areas than the one you mentioned. For one thing, the stereo sub-carrier information is AM sidebands about a center of 38 kHz, and even though the carrier frequency itself is suppressed before transmission, the sidebands, resulting from what started out as an audio-modulated sub-carrier, can only be two in number: one above and one below 38 kHz by an amount equal to the frequency of the modulating tone or note. Thus, even when 15-kHz audio is being impressed upon the L-R channel, sidebands created will be at 23 kHz (lower sideband) and 53 kHz (upper sideband) only.

The two primary reasons for cross-talk that I have discovered relate more to equipment than to spectrum distribution. For one thing, any main channel tuner non-linearities (as for example, in the main channel i.f. strip of the detector circuitry) will cause varying degrees of cross-modulation between main and sub-channels. Remember, too, that main channel demodulated voltages are about *ten times* as great as the amplitude of the 67-kHz sub-carrier which you are trying to work with. Thus, the problem of filtering out main channel is 20 dB greater than if there were a one-to-one relationship.

The SCA decoder which I described in the following issue of AUDIO (October 1968) has an inherent rejection capability (with respect to the main channel) of better than 55 dB, yet, with every tuner with which we've tried it, some deterioration takes place. We did, however, get as high as 50-dB rejection using a good-quality, well-aligned tuner. Incidentally, even the "commercial" units used in restaurants and the like seldom do better than this, but the level is kept low so that the cross-talk is all but inaudible.

We have noted, too, that some stations come through with more of this cross-talk than others, which leads us to believe that station equipment is at fault to some degree, as well. The FCC does not care about setting limits of cross-talk from main channel to sub-channel at the transmitter (since SCA service is not "public programming"), but does set strict limits on SCA cross-talk into main or stereo channels. One of the reasons that SCA operators turn off the sub-carrier during pauses between music is to get around this small amount of cross-talk, which is masked when music is playing on the sub-channel but would be more obvious during periods of silence.

LEONARD FELDMAN
Great Neck, L. I., N. Y.

(Continued on page 98)

A Marantz speaker system is the finest money can buy.

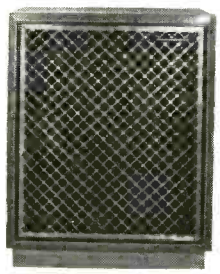
(Our competitors know about this ad.)

Marantz has always set the standards others follow. In preamplifiers. Amplifiers. Turntables. Tuners. And stereophonic receivers.

Today, Marantz once again expands its reputation for audiophonic excellence with the introduction of a new concept in speaker systems.

After years of experimentation, Marantz' first two Imperial Speaker Systems are now ready to be enjoyed by discriminating connoisseurs.

Technically, both feature a three-way design incorporating five speakers. There is a 12" Quad-linear woofer which crosses over at 700 Hz to two mid-range drivers, then crosses over again at 6,000 Hz to two high frequency reproducers.



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.

When you hear, when you see these magnificent speakers, only then can you fully appreciate what goes into making a Marantz a Marantz. Your local franchised Marantz dealer will be pleased to furnish you with complete details and a demonstration. Then let your ears make up your mind.

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EDITOR'S REVIEW

Hi-Fi Show Biz

The 19th IHF-sponsored New York High Fidelity Music Show held in September was a rousing success, according to reports. The IHF advised that public attendance came close to matching last year's 22,500. Just as important, we got the impression that the show-goers were an enthusiastic group who really enjoyed themselves. We were disappointed, however, in the Statler Hilton Hotel's services for the show. Ventilation was very poor, for example, and elevator service was frustratingly slow. Fortunately, spirits were not appreciably dampened. Another unhappy surprise was the poor attendance at the show's free, and heretofore very popular hi-fi seminars. Perhaps the sweltering atmosphere and poor elevator service (seminars were held 14 floors above the exhibit area) were responsible. Also, most show-goers simply forgot about the seminars as they darted from exhibit to exhibit.

West Coasters will also have an opportunity, by the way, to enjoy high-fidelity seminars. The seminars will be featured at the San Francisco show, October 31 to November 3. An interesting innovation at the upcoming San Francisco Show will be a Bonus Savings Certificate to all Show visitors that allows them a saving on component equipment purchases of at least $200. (Fair-traded merchandise or products with special franchise limitations will be excluded.)

Another hi-fi component show—a European one—was presented in September: The "Hi-Fi 68 Dusseldorf." This first hi-fi exhibition in Federal Germany had 124 exhibitors from 12 countries. An official report indicates that 32,000 tickets were sold. The West German show also featured hi-fi talks by experts, as well as musical performances. We understand, too, that exhibit rooms were sound proofed.

Philadelphia will be the scene of a hi-fi component show, February 7-9, at the Benjamin Franklin Hotel. The show will be presented by Almo Radio Co., Inc., with many hi-fi equipment manufacturers participating in it. Almo presented a similar hi-fi show in the spring of '67, as AUDIO readers and many Delaware Valley residents will recall. As in the earlier show, component equipment will be *sold* at the show!

Sound Talk

The 3M Company has revived its quarterly "Sound Talk" bulletins, the first of which has already been published. If the five-page bulletin on magnetic tape is any indication of what is to follow, it should easily equal popularity of the company's earlier bulletins. Readers can obtain these bulletins FREE by writing to the 3M Company, Magnetic Products Div., Marketing Services Department, 3M Center, St. Paul, Minn. 55101.

"We Try Harder"

Avis-Rent-A-Car has a new version of the "We Try Harder" theme in its well-known button. It's a reproduction of each word as it appears visually on an oscilloscope's cathode ray tube (obtained by speaking the words into a microphone).

Electronic Music

New Yorkers and Bostonians can listen to electronic music by Jean Eichelberger Ivey on WNDT-TV (Channel 13, New York) weekdays at 8:55 A.M. and in station breaks scattered through the day. The music, called "Continuous Form," plays on a continuous-loop tape cartridge; excerpts are selected at random, each accompanied by films by Wayne Sourbeer in a random association. The Psyche Sound and Sight is also being used a few nights each week by WGBH-TV (Channel 2, Boston).

The New York Audio Engineering Society Convention and Exhibit, celebrating the AES's 20th anniversary October 21 to 24, has a complete technical session on developments in electronic music systems, with five technical papers on the subject being presented. And it is reported that the new West Coast hi-fi seminars at the San Francisco High Fidelity Music Show will feature, among others, Robert Moog, who will give a demonstration-lecture on electronic music.

Judging from the great activity in this field, as noted above, and with recordings for LP discs and broadcast commercials making wide use of electronic music, interest in this "music" form is not limited to "under-30" listeners. *A.P.S.*

Invitation to euphoria.

PHOTOGRAPH BY FRANZ EDSON

Among all those who listen to music from records, there is a select few who do it very, very seriously. They originally spent countless hours comparing one component against another. Then they tried their speakers here and there at home until they worked to perfection with the room.

And when people like this listen, they do nothing but listen. Just as though they had paid good money for dinner out, orchestra seats and a baby sitter.

They know what that record should sound like. From deep soul-satisfying bass to those delicate, sweet highs.

They're never satisfied until they find themselves in that blissful state that tells them there's just nowhere else to go.

Euphoria.

If you don't know it, just leave everything as it is. Except your cartridge and favorite record. Take both to an audio dealer who has a particularly good listening room.

Listen first with your present cartridge. Then with the golden XV-15/750E. That's all.

You won't mind spending the sixty dollars. It's the least expensive passage to euphoria you'll ever find.

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THE XV-15/750E, WITH A DYNAMIC COUPLING FACTOR OF 750, DELIVERS 100% MUSIC POWER THROUGHOUT THE ENTIRE AUDIBLE RANGE AT 1/2 TO 1 GRAM TRACKING FORCE. IT IS THE NEWEST AND FINEST OF PICKERING'S XV-15 SERIES. EACH XV-15 MODEL IS DCF-RATED FOR A SPECIFIC CALIBER OF TONEARM, FROM A DCF OF 200 TO 750, AND ALL DELIVER 100% MUSIC POWER. PRICED FROM $29.95. DYNAMIC COUPLING FACTOR AND DCF ARE SERVICE MARKS OF PICKERING & CO.

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Model S-8800a $399.50

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Altec Acousta-Voicing*

DON DAVIS

Part 1

How the acoustic characteristics of a room can be compensated by a new method of applying properly designed filters in the sound system

EQUALIZATION OF SOUND SYSTEMS to compensate for the detrimental interaction that can occur between the sound system and the acoustic environment (auditorium, arena, etc.) where it is installed has been under serious investigation by physicists, engineers, and experimenters for a number of years. (See Footnotes 1, 2, 3, 4, 5, 6, and 7.) These investigations determined that the acoustic environment affected the amplitude-frequency response of loudspeakers, as shown in Fig. 1.

William B. Snow, a gifted scientist, showed that a sound reinforcement system near its regenerative state (just below acoustic feedback) could and did increase the decay time of certain highly reverberant frequencies.[8] Finally, Richard V. Waterhouse of The American University, Washington, D. C., applied the classic Nyquist criteria to discuss the conditions necessary to achieve regeneration in a sound reinforcement system.[9,10,11] With these major breakthroughs and the accumulated experience of hundreds of skilled but anonymous workers in the professional sound field, the stage was set for the realization of *detailed* sound system equalization.

Altec Acousta-Voicing

Acousta-Voicing is a total programmed approach to the application of these discoveries to the design, installation, testing, and adjustment of a sound system's amplitude/frequency response to the acoustic environment where it is installed and used.

While the words "Acousta-Voicing" and its derivatives are more than likely due the same eventual obscurantism that finally engulfed the words "high fidelity" and "stereophonic," this two-part series will describe what they mean today while they are still pioneer-pure.

The descriptive term Acousta-Voice* stems from the time-honored practice of voicing or regulating a pipe organ to fit the acoustic environment in which it is installed. Therefore, an Acousta-Voiced sound system is one that has been "voiced" or regulated to fit its acoustic environment. In practice, Acousta-Voicing views the system design in terms of the maximum results that theory will allow, designing up to, but not beyond, that point. It further sets up procedures and tests to ensure the ability of the electronic portion of the system to support the acoustic requirements. Finally, it regulates the amplitude frequency response of the total electro-acoustic system, including electronics, transducers, *and the room itself* to the overall response that allows the greatest tonal quality and acoustic gain.

Acousta-Voicing removes no usable program material, but, rather, brings into equality with the majority of frequencies those special frequencies that the room and sound system together actually over-emphasize. (See Figs. 2A and 2B.) Through the use of a detailed technique and precision test equipment, these adjustments can be made with speed (as little as two hours) and accuracy. The audible results are high acoustic gain, wide and uniform acoustical frequency response, and remarkable freedom from reverberant coloration.

As might be expected, Acousta-Voicing is not for the beginner. The successful Acousta-Voicer must understand the basic design problems of sound reinforcement systems and *how to solve them*. This first article will discuss the requirements for good hearing in a space, the role of the sound reinforcement system, the nature of acoustic feedback, reverberant coloration, and the basic design criteria for Acousta-Voicing.

Requirements for Good Hearing in Auditoriums. Traditionally, four basic factors are considered to determine how satisfactory the listening conditions in an auditorium will be. These four factors are:

1. Quietness of the auditorium.
2. Useful loudness of the program material.
3. Evenness of distribution of program material to all areas in the auditorium where there are listeners.
4. Avoidance of excessive decay time or delay time in the reception of the direct and reflected sound combined at the listener's ears. ("The successive sound in rapidly moving articulation, either of speech or music, is clear and

*T.M. LTV Ling Altec Inc.

[1] Harry Kimball, "Attenuation Equalizers," *Motion Picture Sound Engineering*, D. Van Nostrand Co., Inc. New York 1938. Pages 228-272.

[2] Arthur C. Davis, *"Steps to Improve TV Audio," Video Engineering*, March 1950.

[3] Ercel Harrison, "Hi Q Equaliztion Reactors," Altec Lansing Bulletin AL 1164, 1958. Page 11.

[4] Wayne Rudmose, "Equalization of Sound Systems," *Noise Control* 4, No. 24, July 1948.

[5] C. P. Boner and C. R. Boner, "Minimizing Feedback in Sound Systems and Room Ring Modes with Passive Networks, *J. Acous. Soc. Am.*, 37, 131, 1965.

[6] William K. Connor, "Theoretical and Practical Considerations in the Equalization of Sound Systems," *J.A.E.S.* Vol. 15, No. 2, pages 194-198, April 1967.

[7] Don Davis, "Adjustable ⅓ Octave Band 'Notch' Equalizer for Minimizing Detrimental Interaction Between a Sound System and Its Acoustical Environment," paper given at the 102nd SMPTE Technical Conference, Sept. 22, 1967.

[8] William B. Snow, "Frequency Characteristics of a Sound Reinforcement System," *Journal of the Audio Engineering Society*, Vol. 3, #2, April 1955, pages 74-76.

[9] Richard V. Waterhouse, "Theory of Howlback in Reverberant Rooms," *Journal Acoustical Society of America* 37, 921, May 1965.

[10] H. Nyquist, "Regeneration Theory," *Bell System Technical Journal*, 11, 126-147, 1932.

[11] H. W. Bode, "Network Analysis and Feedback Amplifier Design," D. Van Nostrand Co., Inc. New York 1959.

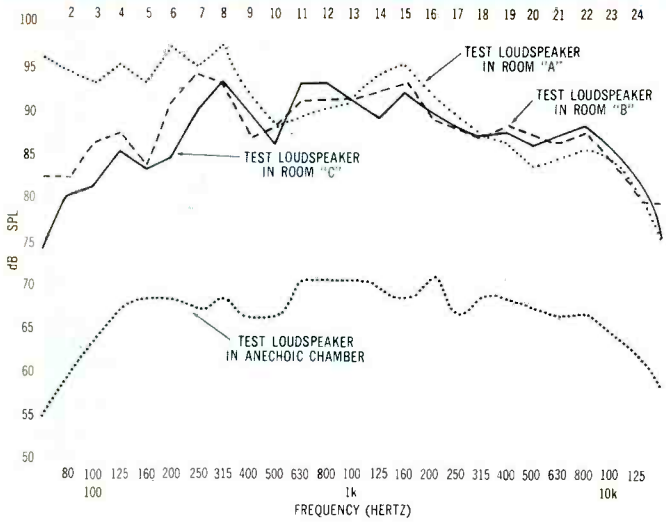


Fig. 1—A comparison of the same loudspeaker in four different environments showing the effects of room size, shape, absorption, and speaker placement on the final response curve.

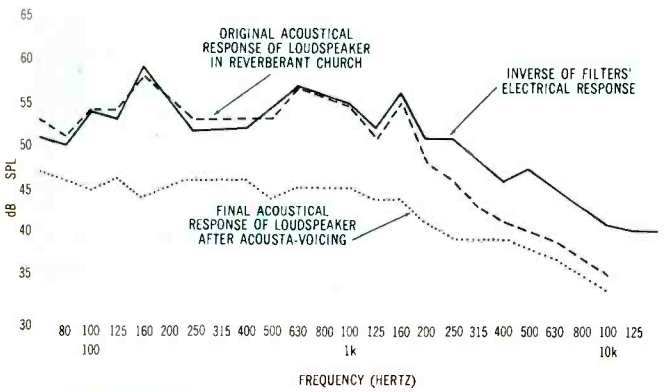


Fig. 2A—Correction of the reverberant field response, with allowance made for normal air absorption, by Acousta-Voicing filters. Also included is a comparison of the filters' inverse electrical response with the original uncorrected acoustic response.

distinct, free from each other and from extraneous noises."[12])

Nature of Acoustic Feedback. The basic theory of amplification[9,10,11] would lead us to expect that the maximum acoustic gain (loudness) we could achieve with any sound reinforcement system would be as follows: Assume that a person talking two feet from an open microphone is generating at the microphone diaphragm 70 db SPL, and that a loudspeaker is mounted 30 feet above the microphone diaphragm. Further, assume that free-field conditions exist. What maximum SPL can we expect to generate at 4 feet from the loudspeaker (the distance at which professional loudspeakers are rated for efficiency) before acoustic feedback should occur.

Naturally we would expect feedback to occur whenever the sound from the loudspeaker reached the same SPL at the microphone's diaphragm as the talker's voice, inasmuch as that represents unity gain. Under the conditions described, the SPL from the loudspeaker could be raised as high as the SPL at the microphone diaphragm from the talker's voice (70 dB SPL). Add to this the inverse-square-law loss of the SPL at 4 feet from the loudspeaker (sound pressure level decreases 6 dB for every doubling of the distance), compared to the SPL at 30 ft. from the loudspeaker: $\frac{30}{4} = 7.5$. Thus, $20 \log_{10} 7.5 = 17.5$ dB. Therefore, we would expect to be able to approach 70 + 17.5 dB SPL at 4 feet from the loudspeaker, or 87.5 dB.

To a person sitting 80 ft. away from both the talker and the loudspeaker, we would expect the talker's voice unamplified to arrive at his ears with an SPL of 38dB SPL at the listener's ears. ($80/2 = 40$; $20 \log_{10} 40 = 32$ dB; $70 - 32 = 38$ dB) The sound from the loudspeaker would arrive at the listener's ears with an SPL of 61.5 dB. ($80/4 = 20$; $20 \log_{10} 20 = 26$ dB; $87.5 - 26 = 61.5$ dB) This would mean that the acoustic gain (61.5 dB − 38 dB) would be 23.5 dB SPL at 80 feet.

A formula for calculating potential acoustic gain has been devised[13]. This formula,

$$\text{S.G.B.F.} = 20 \log_{10} \left[\frac{D_1}{D_s} \cdot \frac{D_0}{D_2} \right],$$

works out in our example as:

D_1 = 30′ distance from microphone to loudspeaker

D_s = 2′ distance from talker to microphone

D_0 = 80′ distance from talker to listener

D_2 = 80′ distance from loudspeaker to listener.

Therefore, $20 \log_{10} \left[\frac{30}{2} \cdot \frac{80}{80} \right] =$ 23.5 dB SPL.

It is theoretically possible to exceed unity gain slightly if the phase angle is manipulated so as to ensure that no amplitude exceeding unity gain passing through the entire system, including the room, has a phase angle equal to 2N radians (N = 0, 1, 2, etc.). By inference, then, feedback that occurs before reaching this theoretically practical limit is due either to a peak in the amplitude response of the combined electronics, the electro-acoustic transducers, and the acoustic environment, or to a detrimental phase relationship between all these components at the feedback frequency.

To further complicate the basic nature of the problem, electro-acoustic transducers have resonance points, and the acoustic environment in a large hall has thousands of normal modes in the usable audio spectrum.

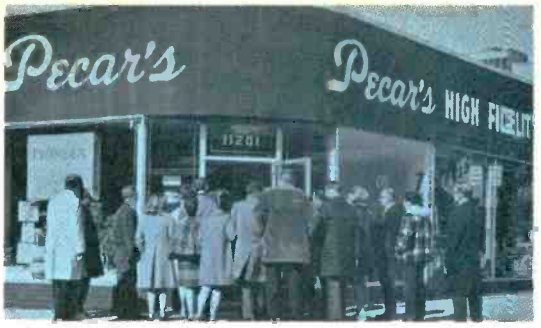
Acoustic feedback, which is the product of the auditorium's high-Q modes in combination, plus the electro-acoustic transducer's response, plus the coupling of the two systems together, can on occasion reduce the actual realized acoustic gain from +23.5 dB to −2 dB or less. When this occurs, the auditorium in question rapidly begins the expansion of its reputation of "having trouble with sound."

It is demonstrable that the sound system in such a room always feeds back at one frequency at a time. But this is by no means proof that only one room mode is involved. Quite the contrary. In a large auditorium, the normal modes overlap significantly, and the response at

[12] Wallace Clement Sabine, *Collected Papers on Acoustics*, Harvard University Press, Cambridge, Mass., 1922, page 4.

[13] Arthur C. Davis and Don Davis, "Microphones for Sound Reinforcement Systems," AUDIO, Dec. 1967, page 65.

the crowd pleaser...



In hundreds of Pioneer franchised high fidelity dealers across the country, the SX-1500T is drawing enthusiastic attention because it is a no-compromise receiver. Its highly sensitive front end pulls in the most difficult stations . . . and is consequently pulling in the crowds. The SX-1500T was made for the thousands who wanted the finest receiver possible . . . at a reasonable price.

The specifications and quality of the SX-1500T are substantiated by its performance and, more importantly, its sound. It boasts an output of 170 watts of music power, an extraordinary capture ratio of 1 dB, a signal-to-noise ratio of 65 dB, and harmonic distortion actually below 0.1% at half rated power (0.5% at full rated power). FM sensitivity is outstanding at 1.7 uv. Frequency response is 20 to 70,000 Hz ± 1 dB.

If you want a better receiver, *don't be misled* — pick the one with the honest price. You owe it to yourself to compare the SX-1500T with any other receiver on the market regardless of price.

See and hear the SX-1500T now. Or write for literature and name of nearest dealer.
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PIONEER'S NEW SX-1500T AM-FM STEREO RECEIVER
170 WATTS, FET FRONT END, AND 4 IC's

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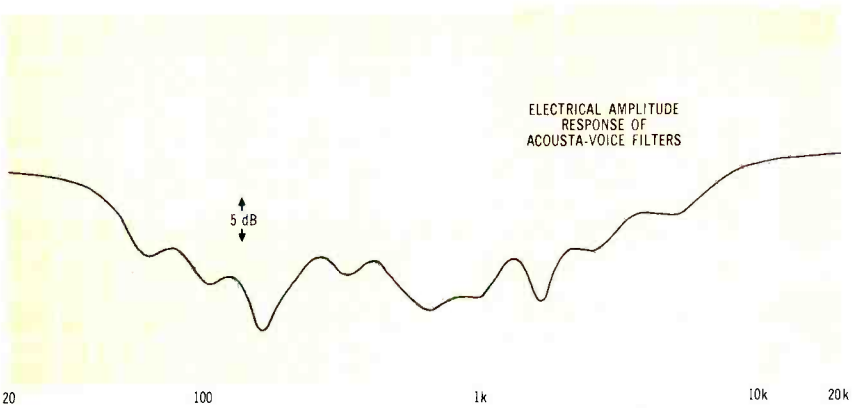


Fig. 2B—This shows the electrical response of the Acousta-Voicing filters in the case of the reverberant church shown in Fig. 2A.

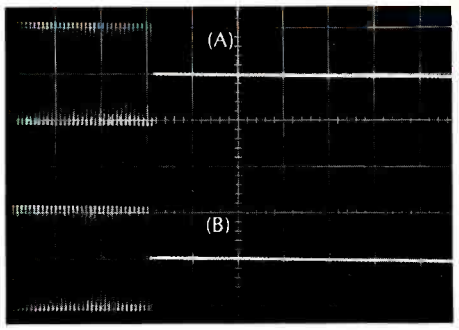


Fig 3. – Tone-burst photographs of the transient response of critical-bandwidth band-rejection filters vs. narrower-bandwidth filters. (A) Source—32 cycles of 1000 Hz signals. (B) signal at output of link circuit.

any one frequency is dependent on more than one mode.

Again, while it can be demonstrated experimentally that appreciable nulls exist between modes, their close spacing and great similarity dictate that they be treated as groups. In a very practical sense, any attempt to handle a high-Q, single room mode requires a very high-Q filter which, due to its high Q, introduces a transient problem in the response of the sound system sometimes equal to or greater than the room problem it attempts to correct. (See Fig. 3.) In extreme cases, sound systems using such filters can be found that exhibit feedback *without a microphone being connected.* As the gain controls are advanced, the sound system goes into feedback as if a microphone were present. Removal of the non-constant k high-Q filters immediately remedies this phenomenon. The problem is aggravated by the impedance of the transmission link suddenly changing at each of the resonant frequencies of the filters.

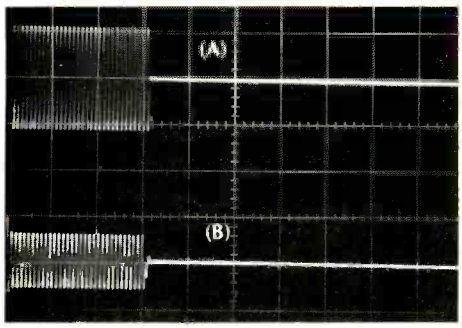
Fortunately, it can again be experimentally demonstrated that controlling the overall amplitude and phase combinations of the electronics, electro-acoustic transducers, and the acoustic environment, with band rejection filters of approximately critical bandwidths, results in greater gain and smoother system response than that achieved by band-rejection filters with narrower bandwidth characteristics. Our ears detect subjective loudness changes in typical program material (music and speech) in critical bandwidths.[14] Making filters narrower than these critical bandwidths at any given frequency results in no audible improvement when compared to the critical bandwidth filter. Since the Q of a passive filter may be measured by its "ringing" (transient decay distortion) it is of real interest to determine how narrow in bandwidth a band rejection filter can be made and still pass undistorted transients.[15] Happily, "ringing" does not become a problem until the filters are made narrower than the desired critical bandwidths. Additionally, the band-rejection filters of approximately critical bandwidth can be built as constant k in circuit configuration, thereby avoiding the aforementioned transient distortion when they are introduced into the electronic portion of the system. (Early experimenters ignored this "ringing" problem, but professional sound contractors, motion picture, recording, and broadcast engineers cannot.)

Reverberant Coloration. The second major limitation encountered in large auditoriums with well-designed and properly installed sound systems is a reverberation decay time that jumbles and garbles the amplified speech. By subjective tests of speech intelligibility, followed by objective measurements of the length of time given tones take to decay in various auditoria, it was determined over the years within what parameters of decay times that speech intelligibility was affected. In a definitive paper given to the Audio Engineering Society in 1955[8], William B. Snow described and demonstrated the increased decay time that resulted if the sound system operating near regeneration picked up the natural room decay. It became apparent from Mr. Snow's data that, in the attempt to raise the signal-to-noise ratio via a sound amplification system, it also could increase the decay time, at selected frequencies, by as much as six to seven times their original decay time. (See Fig. 4.) Thus, in the attempt to get the direct sound louder, the sound system, by operating close to feedback, also made the re-amplified decay time longer.

Mr. Snow's paper also showed that, as the sound system was operated well below regeneration (−10 to −12 dB), it no longer amplified the natural decay time of the room. As shown earlier, if the acoustic response of a sound system is made more uniform, acoustic gain increases towards its theoretical limit. In so doing, the sound system then has sufficient acoustic gain developed to allow it to be operated well below the feedback point, thus eliminating the magnification of the natural reverberant decay times in the room.

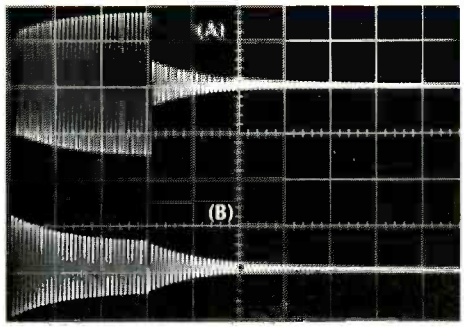
When a sound system has been equalized to produce the most uniform amplitude response possible, as measured in the reverberant far field, there is considerable improvement in acoustic quality and gain. The practical results to the user are manifested as complete freedom

[14] Wolfgang E. Ohme, "Loudness Evaluation," Hewlett Packard Journal, Nov. 1967. Pages 6, 7, and 10.
[15] General Radio Co. "Q of a Resonant Circuit." Publication #STX-102, Oct. 1966.



(A) Source – 32 cycles of 1000 Hz signal.

(B) Link circuit with constant-k filter inserted and set for —6 dB.



(A) Source – 32 cycles of 1000-Hz signal. (Source pulled into ringing by filter.)

(B) Link circuit with series-tuned inductor filter inserted and set for —6 dB.

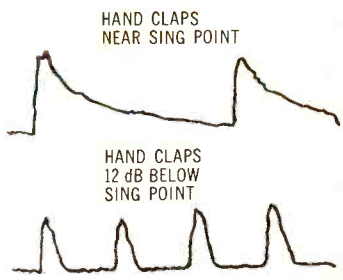


Fig. 4—The effect of an amplification system, near regeneration, on the decay time of a handclap in a reverberant space compared to the same system well below the regeneration level.

Courtesy of William B. Snow.

from feedback—it can now be operated well below regeneration, as excess gain is available—and causes minimum excitation of the room's natural reverberation. Sound systems have been Acousta-Voiced in spaces with reverberation times as high as 10 sec. at 512 Hz! The actual in-the-field results often seem miraculous to the first time observer. But it is demonstrable over and over.

Basic Steps in Acousta-Voicing Process. Because of the inextricable dependance of successful sound-system equalization on the quality of the sound system itself, it must be recognized that Acousta-Voicing, in order to succeed, must encompass more than the mere application of filter circuits to an existing sound system.

Altec Acousta-Voicing contractors, for example, follow a step-by-step method that ensures that no vital link in the total chain of interacting components is overlooked. These steps are:

1. Design a new system or re-design an old system of either the central high-level or overhead distributed type (or necessary combination of the two that ensures sufficient electrical power to generate the required acoustic power at the listener's ears, and that guarantees even distribution without "hot" or "dead" spots. This automatically precludes re-entrant horns, sound columns, loudspeakers placed on both sides of a stage, loudspeakers mounted in strings down side walls, and other self-defeating but time-honored practices.

2. Perform complete tests of the installed systems using accurate calibrated test equipment. Tests should include, but are not restricted to, inspection of:

(A) Proper grounding of all components and circuits to a real earth ground.

(B) Link circuits and loudspeaker-line impedances and their adjustment to optimum values.

(C) Hum, noise, spurious oscillations and crosstalk, and their elimination, if present.

(D) Acoustic distribution of the loudspeakers' output throughout the audience area.

(E) Time-delay relationships to ensure that the time relationship between the live talker's sound and the amplified sound is sufficiently short.

(F) Electrical levels to avoid an early stage either being overdriven or overdriving a subsequent stage.

3. After designing, installing and testing the sound system, and only when each of these steps is found to have been properly accomplished, can the adjustment of the sound system's acoustic response in the room environment begin.

It is this third stage that is popularly identified with Acousta-Voicing. Because the interested observer, such as the owner of a building with a long acoustic tale of woe, observes that his building ceases to be a problem to him, he not unnaturally assumes the "miracle" lies in the Acousta-Voice* filters alone. Word of such success spreads quickly, of course. But unfortunately, many experimenters with equalizers are disappointed in their results. It should be noted that when all three of the foregoing steps are properly executed, it is our experience that Acousta-Voicing *always works*. But it cannot be performed by untrained sound-system designers, engineers without proper precision test equipment, or sound contractors without access to both quality components and equalizers. This third step breaks down into three parts:

(A) Broadband equalization of the overall acoustic response of the system in the room.

(B) Narrowband treatment of the feedback frequencies that the room-sound system combination prefers.

(C) The control of feedback triggered by proximity effects exhibited by the microphones.

4. The final step is usually carried out by the user to suit his own tastes. This consists of varying the overall broadband acoustic response of the sound system to best please his or his customer's ears. In some cases, he may prefer to sacrifice acoustic gain for an acoustic effect beneficial to the program material. He should be provided in all cases with the necessary program equalizers to allow reasonable experimentation, but always of the type that allows instant return to the calibrated base as set up by the Acousta-Voice* engineer.

The second part of the series will discuss the design, installation, and the actual Acousta-Voicing of a sound system.

CONCLUDED NEXT MONTH

AUDIO'S Guide to the

1968 San Francisco High Fidelity Music Show

Civic Auditorium,
99 Grove St., San Francisco, Calif.

TIMETABLE:

October 31 (Thursday)
4:00 pm to 10:30 pm

November 1 (Friday)
4:00 pm to 10:30 pm

November 2 (Saturday)
1:00 pm to 10:30 pm

November 3 (Sunday)
1:00 pm to 9:00 pm

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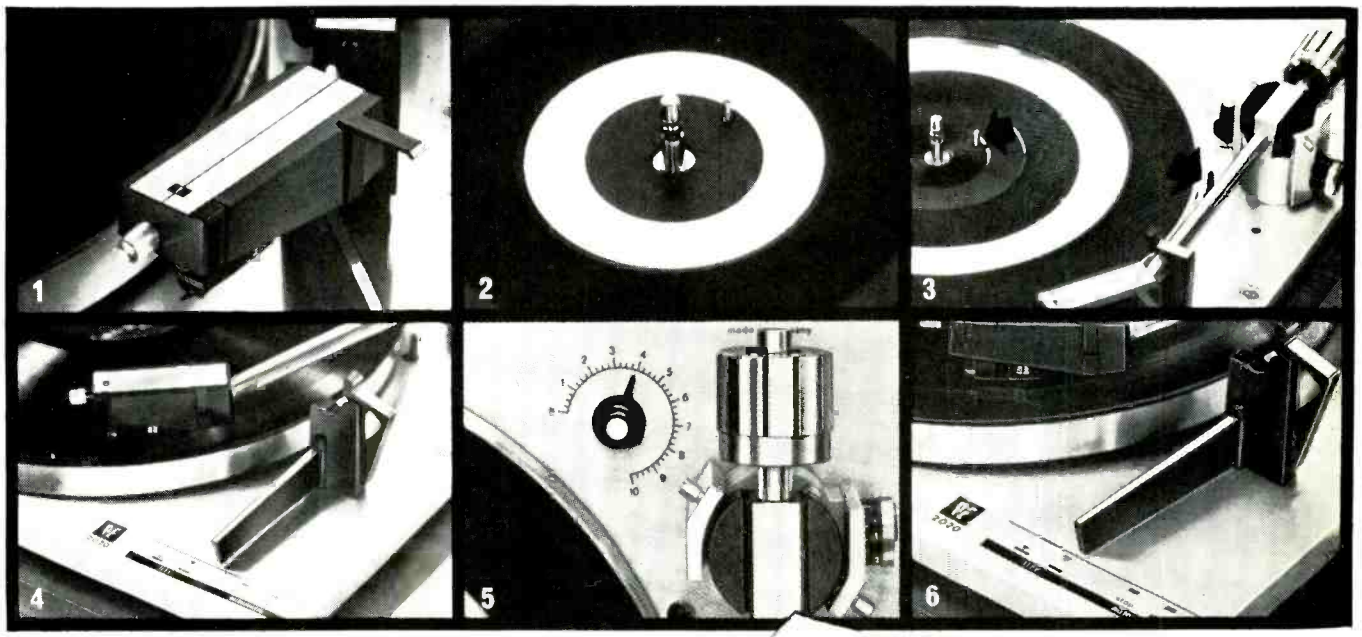
JUDGING FROM THE exciting new products displayed at the New York High Fidelity Music Show last September, the upcoming San Francisco show should be equally attractive to show goers.

Some manufacturers made extra-special efforts at the N. Y. show to create attractive exhibit rooms. From a decor viewpoint, three exhibits captured our attention: *Stanton*, with two rows of modern-style, leatherette chairs, each with stereo headphones mounted on what appeared to be Luxor desk lamp swivels. Simply sit down and swing the headphones over. Good idea for the home. *Pioneer* had a lovely Japanese decor, Shoji screens and all, complementing its exhibit rooms. *Pickering* showed a variety of antique instruments, together with historical information. Other interesting exhibits included *Garrard's* SL-95 being operated with voltage varying from 65 V to 125 V, while an electronic counter verified that motor speed did not change. *Crown's* DC-300 amplifier was shown lighting up four 150-watt bulbs with 60 Hz fed into the amp's input. Catch 'em all at the West Coast Hi-Fi Show, October 31 to November 1.

Here are some ear-catchers to look for that were not mentioned in AUDIO's Hi-Fi Show preview last month: *CM Labs* packed 'em into its N. Y. exhibit room with a $1050 (yes, that's four figures) stereo FM tuner that features digital readout of frequencies on Nixie tubes (otherwise the tuner is fully solid-state). It has 100 crystal-controlled frequencies, so there's no drift. Press a button and the station clicks in. There's a timer/programmer and a remote-control option, too. *Bogen* and *Fisher* both displayed Varactor-tuning receivers, each with a low-cost optional remote-control device (connected by cable) for selecting broadcast stations from a seat at the other end of the room. The *Fisher* model (Model 500TX AM/FM stereo at $449.95) utilizes dual-gate MOS devices in its front end (as does the *CM Labs* stereo tuner mentioned previously). To our knowledge, these are the first consumer uses of this advanced transistor type. Show attendees visiting *Panasonic's* exhibit could be seen trying out the company's new headphones — with a complete battery-operated FM stereo receiver mounted inside. Price is expected to be under $100. *H. H. Scott* spotlighted its new "Designer Stereo" AM/FM stereo/phono "consolette," which also has LP-storage space. *Sherwood* showed a compact for the first time, its Model S-6000. The 100 watt, AM/FM stereo compact gives the user a choice of turntables from among *Garrards*, *Duals* or *Sherwood's* own SEL-100 (a photo-electric actuated turntable with piano-key controls). Price is $419.50 plus speakers and turntable. *Superscope* showed its Model 630 stereo tape recorders, a $379.50 unit with three tape heads, echo and sound-on-sound facilities, fader-type level controls, and a noise elimination circuit. *Dynaco* entered the loudspeaker market, displaying a bookshelf speaker system imported from Europe. Containing two drivers, it is expected to retail for about $80. *Elite* showed small air-suspension speaker systems, "Sound N' Color," with decorator-color enclosures. Priced at $24.95. *Marantz* expanded its line with a lower-priced FM stereo tuner (Model 20 at $395.00) and a lower-priced AM/FM stereo receiver (Model 19 at $495.00). Æ

the new ELPA PE-2020 Automatic turntable lets you

escape from the ordinary



Here's why

(1) The Exclusive 15° Vertical Tracking Angle Adjustment. For critical listening and perfect sound reproduction, records should be played with the stylus at a 15° vertical tracking angle. The new ELPA PE-2020 is the only automatic turntable that permits the critical listener to do this — for a single record, in single manual play . . . or for any record in a stack in multiple automatic play. This feature gives the ELPA PE-2020 the precision of a fine manual turntable, and a greater precision in multiple play than any other automatic turntable.

(2) Stylus Protection. It is impossible to damage the stylus of the ELPA PE-2020 by lowering the tonearm onto an empty platter. Should the turntable be switched on accidentally, the tonearm will refuse to descend if no record is on the platter.

(3) Automatic Scanning. You don't need to adjust the new ELPA PE-2020 for various size records. The scanning device **automatically** determines the size of the first record on the platter and **automatically** adjusts the tonearm to descend in the proper play position.

(4) Simplicity Of Operation. One lever controls all modes of operation: Start, Stop, Repeat, Cueing, Pause, and Lift — making the ELPA PE-2020 the easiest automatic turntable to operate. The single control is located at the front of the turntable and is easily accessible even in confined quarters.

(5) Anti-Skating. The most sensitive anti-skating device on any automatic turntable. Combined with an exact adjustment dial to compensate for stylus shape, size, and tracking weight. Less wear on your records, more perfect sound reproduction.

(6) Motor Driven Cueing. The most advanced form of cueing today. No extra levers, no viscous-damped hand controlled manual devices. Eliminates accidental slips of hand striking the tonearm.

AND THERE ARE MANY, MANY MORE SUPERLATIVE FEATURES ON THE NEW ELPA PE-2020.

Don't make a buying decision on an automatic turntable without seeing the finest . . . the new ELPA PE-2020. See it at your high fidelity dealer, or write for full literature and name of nearest franchised dealer. ELPA MARKETING INDUSTRIES, INC. • New Hyde Park, N.Y. 11040

Audio Engineering Society Celebrates 20th Anniversary

C G. McPROUD

C. J. LeBEL—first president of the Society (1949), secretary, 1952-1965. Vice-President, Audio Devices, Inc., Chief Engineer, Audio Instrument Company, and formerly Chief Engineer of The Maico Company, hearing aid manufacturer. "C.J." was the guiding genius of the Society, and a true audio pioneer who firmly believed that audio engineering was a profession separate from radio or electronic engineering.

THE THIRTY-FIFTH CONVENTION of the AES marks the 20th Anniversary of the founding of the Society. For the last ten of these years, the Society's activities have been well publicized; its early history, however, is less familiar. AUDIO Magazine is proud of its part in the formation of the Society, since it was in these pages that the following letter appeared in December, 1947, eight months after the magazine began its life, as *Audio Engineering*:

Audio Association

Sir:

After receiving the first few issues of your new magazine, I must say that I believe you are serving a much neglected field, in a highly adequate manner.

Now that the audio engineer has been dignified by a specialized publication, which will tend to draw the members of the field together, is it not time for him to have an organization of his own?

I have in mind an association similar in function and purpose to the I. R. E. and S. M. P. E. in their respective fields.

I will be glad to correspond with anyone interested in this matter.

Frank E. Sherry, Jr.
705½ W. San Antonio St.
Victoria, Texas

What do our readers think?—Ed.

One reader, at least, did something about it, as is evidenced by his reply in the succeeding issue, January, 1948:

Sir:

In the last issue, Mr. Frank E. Sherry, Jr., suggested that audio engineering had grown to the point where it needed a professional society of its own.

A group of us, long active in broadcasting and recording, feel the same way he does. Audio engineering will be unhampered only when it has a society devoted exclusively to its needs—controlled by, and run only to benefit, the audio engineer.

We have been discussing this matter for several months, and are preparing to hold an organization meeting.

Will those interested in such a society please write the undersigned, giving the following information:

Name
Mailing Address
Company
Title or nature of work

We will notify you of the meeting date.

C. J. LeBel
307 Riverside Drive
New York 25, N. Y.

Obviously there had been some previous discussion of the need for such a society in the New York area, and many audio men responded with encouraging zeal. The first meeting of audio engineers was called for January 8, 1948, and a steering committee consisting of C. J. LeBel, John D. Colvin, Norman C. Pickering, Chester A. Rackey, and the writer, was appointed. The organization meeting was held on February 17, 1948 at the RCA Victor Studios on East 24th Street, with 137 persons attending.

Among the reasons for forming the Society was the need to establish audio engineering as a separate profession. Also desirable was the provision of a means for publishing papers on the subject.

Even before the Constitution was drawn up, the newly formed Society began to hold technical meetings—the first on March 11, 1948, at which Dr. Harry F. Olson gave a talk on "Some Problems of High Fidelity Reproduction," with demonstrations.

Through correspondence with audio engineers throughout the country, Mr. LeBel began to get responses in other cities, and sections were organized in Denver, Hollywood, and San Francisco.

The first slate of officers was elected September 28, 1948, and the Society was finally officially in operation, with C. J. LeBel as President, Chester A. Rackey as Vice-President, Norman C. Pickering as Secretary, and Ralph A. Schlegel as Treasurer—a post he holds to this day. Mr. Pickering remained as Secretary for three years, and his post was then taken up by Mr. LeBel, who held it until his death on April 13, 1965. It was always "C.J.'s" function to keep things moving, and he did this well, devoting his time indefatigably to the Society's activities.

The First Annual Convention was held in conjunction with the Audio Fair in October, 1949, at the Hotel New Yorker, and the first West Coast Convention was held in 1954. Since then, the Society has held two conventions each year. The Society's first technical papers were published in the AES section of *Audio Engineering* until January, 1953, when the *Journal of the Audio Engineering Society*, now a quarterly association publication, first made its appearance. AES membership now numbers some 4000, and active sections operate in New York, the Midwest, Washington, Los Angeles, San Francisco, and Tokyo. Æ

Some of the driving forces behind the AES's growth

JERRY B. MINTER—sixth president. Vice-President, Measurements Corporation, Boonton, N.J., and President of Components Corporation, Denville, N.J. Fellow and past president of the Radio Club of America, and a member of the Aircraft Owners and Pilots Association.

SHERMAN FAIRCHILD—tenth president. Mr. Fairchild is president of Fairchild Recording Equipment Co., and active in a group of companies bearing his name. His financial aid to the Society was instrumental in getting the Journal started, and in providing the preprints of convention papers.

ALBERT A. PULLEY—seventh president and one of the original Governors. Manager, general recording, RCA Victor Record Division, New York.

DR. HARRY F. OLSON—thirteenth president. Director, David Sarnoff Research Laboratories, author of several books on music and acoustics. Dr. Olson was the first recipient of the John H. Potts Memorial Award.

COL. RICHARD H. RANGER—eighth president, and behind him, F. Sumner Hall, fifth president, at an early AES banquet. Colonel Ranger was president of Rangertone, Inc., manufacturers of tape recording equipment, and his work on sync systems for tape recording is the basis for many present systems. Mr. Hall was President of Audio Equipment Sales, manufacturer of accessories for studio use.

HERMAN HOSMER SCOTT—fourteenth president. President H. H. Scott, Inc., manufacturer of high fidelity equipment, sound level meters, and similar laboratory instruments. Mr. Scott was the third recipient of the Potts Award.

WALTER O. STANTON—ninth president. President of Pickering & Co. Inc., and of Stanton Magnetics, both of Plainview, N.Y.

DR. JOHN K. HILLIARD—one of the original Governors of the Society, and the thirteenth recipient of the John H. Potts Award. Dr. Hilliard is Director of Engineering, Ling-Altec, Anaheim, California, and formerly with the sound department of Metro-Goldwyn Mayer.

What the Pro's expect in the next 20 years

When AUDIO celebrated its 20th anniversary last year, we asked a number of the industry's executives what they thought we could expect in high fidelity in the next 20 years, and they responded with their prognostications. Now that the Audio Engineering Society is celebrating its 20th anniversary, we thought that a similar set of prognostications from leaders in the professional field would be enlightening. We asked, and they replied with the following estimates, guesses, hopes, expectations, or PROGNOSTICATIONS

LEO L. BERANEK
Bolt Beranek and Newman Inc.
President, Audio Engineering Society

Loudspeakers are the weak link in the high fidelity chain. To produce clean bass at 40 Hz and 100 decibels in a living room, the necessary magnetic and radiating structures cause a displacement of a volume of air of about 500 cm^3 on peaks, which, for a 10-inch diameter cone, corresponds to a linear peak-to-peak excursion of 3 cm. Even if perfect electroacoustic conversion were accomplished, the standing-wave pattern in the room would mess it up. This much loudspeaker displacement is required simply to move the eardrum about 0.03 cm, corresponding to a maximum volume displacement of only one-hundredth of the loudspeaker cone.

I prognosticate that by 1980 we shall have microminature batteries—a development that will permit the hi-fi listener to plug two tiny radio receivers into his ears. Thus, the glories of perfect stereo reproduction will be his, with flat response, no distorion, and full loudness. Architects and wives would cheer the new freedom from living-room clutter. Assembly-line workers using the hi-fi plugs could enjoy work-with-music that would be free of the din and distractions of the plant.

Stereo plugs could also be designed to pick up acoustic stimuli simultaneously so that the listener could use music for background and yet remain in normal acoustic contact with his surroundings. Even the telephone receiver could incorporate electromagnetic shielding that blocks the background music when the receiver is pressed against the ear.

BENJAMIN B. BAUER
CBS Laboratories

This time, instead of making prognostications about equipment and techniques, I wish to address myself to the men and women who will engineer and operate the world of sound.

You will witness (to paraphrase McLuhan) the continued implosion of our electronic space-time. What we think of as being difficult today, by comparison, will loom simple tomorrow as fresh complexities emerge to encompass the new interactions of the medium with the message, sight and sound, tactile (as in Burris-Meyer's infrasonic theatre sound), olfactory, gustatory, and perhaps even extrasensory influences. So you will have to know more and think new to stay on top. In the antiquity of 1967 one could design a saleable loudspeaker, through inspired "putzing" with random sample cones and magnets. In 1968 and beyond you will need to use holographic interferometry and write a computer program to cure the cone anomalies; or better yet—forget the cone and use modulated ozone or something.

A new engineering philosophy more-and-more will be needed: More care with the time-space of experiments, more attention to reliability and economy, more human engineering (for the maker, the user, and the repairer), more interaction between the scientist and the artist through quieter machines, better soundproofing, and better earmuffs—in short, more personal involvement in the fields where, at any given point along the time coordinate, the human excels and the machine falters.

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.

STANTON

Check No. 31 on Reader Service Card

Prognostications

LOU BURROUGHS
Electro-Voice, Inc.

Twenty years ago I made a prediction that the recipe for success in the acoustic business was dedication to reproduction of flat sound in every link in the chain.

Our dedication to this concept over the past twenty years has reinforced my belief and that of my associates that the function of the manufacturer in the electro acoustic business is to provide the user with the mirror image of the original sound.

As we have strived to achieve this, it became increasingly obvious that to reach our objectives in one element within the chain and not in another would negate a great deal of the user benefits. In the years ahead, we believe firmly that loudspeakers, microphones of all types, and of course the electronics linking the pick-up and reproduction will be improved in terms of versatility, flatness, reliability, and ease in handling.

Revolutionary changes did not take place in the past two decades, and are not likely to occur in the next two. The gains made in acoustic products have been painstakingly achieved by the combination of more sophisticated materials, better instrumentation, and of course the incalculable aid of scientific electronic data processing equipment, which has made it feasible to perform years of calculations in hours.

As true high fidelity, which might better be described as natural fidelity, becomes easier to obtain, it will be more widely sought until every form of communication may well bask in its subjective benefits. The reproduction of music, the reinforcement of sound, intercommunication, indeed all forms of audio transmission, will become better as a result of the continuing advances made because of the need of the professional.

JOHN M. EARGLE
RCA Record Division

There is enough that we *know* will happen in the next twenty years without having to speculate upon what *might* happen. Here are some thoughts which occur to me:

We will continue to break records as we have in the past twenty years. In electronic components this means better performance in less space and for less money. Miniaturization need not continue beyond a certain practical limit. However, tape and disc speeds will likely continue to diminish as new developments make it possible to hold onto, or even better, our current quality standards.

One of the "last frontiers" in audio is the matching of loudspeakers to listening rooms. A year and a half ago there was virtually no activity in this area. This year it is a big business in sound reinforcement as well as reproduction in the home. There are already loudspeakers for sale with "black boxes" that match room-loudspeaker characteristics, and these are destined to become far more sophisticated in the near future. We will also see an increase in bi-amplification as well as tri-amplification of audio channels as part of a return to more sophistication on the part of tomorrow's audiophiles.

Today's techniques in noise reduction will find their way into tomorrow's products. This will come first to recorded tape and then to discs. These techniques will result in recorded product in the home with signal-to-noise improvements of 10 to 15 decibels, and one of the consequences of this will be excellent quality on tape at 3¾ ips and superb quality at 7½ ips. The heart of the system would be a small unit, a special kind of pre-amp, which would be switched into the circuit to "decode" and restore to normal tapes or records that have been processed from specially compressed masters.

Noise reduction techniques will make more practical a move toward multi-channel playback in the home. Just as two-channel stereo provided a leap forward in realism over mono, so multi-channel techniques—four and even beyond—will again carry us forward. Few people have had the privilege of hearing a four-channel playback with reverberant information separately recorded. The effect is awesome, and it truly enables us to recreate in our living rooms the ambience of much larger spaces, from recital halls to cathedrals.

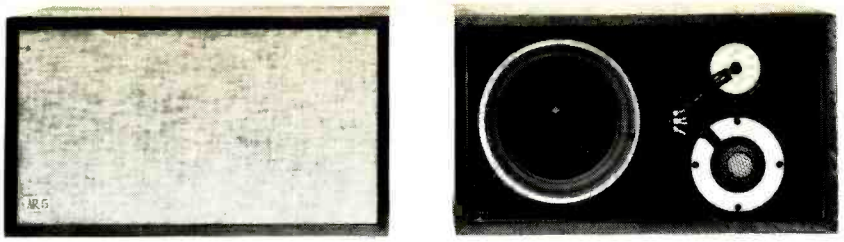
In the recording studio certain changes are inevitable. Even more so than today, pop music will be created in the control room, and the sky will be the limit as far as the use of signal processing equipment is concerned. Automation, currently a reality in the areas of master tape and lacquer transfer, will undoubtedly move into other studio areas.

Finally, there is the one development that we are all waiting for and that is Sight and Sound, or whatever you want to call it. Whether it is on film, tape, or disc, we will have in our homes video and audio on the same medium. Research in this area is proceeding at quite a pace, and one dares not hazard a guess as to which method will win out.

IRVING JOEL
Capitol Records, Inc.

Twenty years ago, five 12-inch 78-rpm records sold for $7.50. Today the same amount of music is recorded on a single stereo LP and lists for $6.00.

The record-making process is ex-



the AR-5

a new addition to the AR family of speaker systems

In October, 1967, after nine years of experimentation and development, Acoustic Research introduced the AR-3a speaker system. It is the best speaker system we know how to make, regardless of price. The most important innovations in the AR-3a are two new hemispherical speakers which provide very smooth mid- and high-frequency response, together with what one reviewer called "virtually perfect dispersion." These two hemispherical speakers have now been combined with an entirely new 10-inch woofer to make the AR-5, a speaker system almost as good as the AR-3a at a price about $75 lower. The main difference between the two systems is that the AR-3a response extends approximately one-third octave lower.

The cone of the AR-5 woofer is molded by a new low-vacuum process developed especially for Acoustic Research. The unusual cone texture which results reduces greatly the tendency toward coloration heard in conventional molded cones of paper or polystyrene. At the cone's outer edge is a new suspension, molded of urethane polymer. The cone itself has a compound curvature which is new, it is in a new housing, and the voice coil attached to it is slightly larger and longer. These internal improvements are complemented by a low 650 Hz crossover frequency made possible by the wide range of the AR hemisphere used for mid-frequencies. The crossover network is of the same type as is used in the AR-3a, and uses 100 mfd of highly reliable paper-dielectric capacitors. The two level controls are fully compatible with transistor amplifiers at all settings, as are the controls of all AR speaker systems.

The AR-5 is priced from $156 to $175, depending on cabinet finish, and is exactly the same size as the AR-2x and AR-2ax: 13½" x 24" x 11½" deep. Impedance: 8 ohms.

Please write to us for technical data and descriptive literature.

ACOUSTIC RESEARCH, INC., 24 Thorndike Street, Cambridge, Mass. 02141

Overseas Inquiries: Write to **AR International** at above address

Check No. 33 on Reader Service Card

Prognostications

tremely complicated and requires many steps between the sound pick-up and the finished record, especially since somewhere near the middle of the process we have to cut a lacquer master with a "hot dull chisel" and play the finished record with a "round ball."

Where are we going? Pictures on disc? It's already here, yes. They're black and white and still, but the possibilities of color and motion are well within sight. In fact, there is a magnetic disc which will reproduce color.

Tape, after a long awkward adolescence, is finally showing signs of becoming an excellent recording medium. New tape formulas, high-quality, high-speed duplication, and new packaging techniques will strengthen the usefulness of tape in the record industry.

I feel that in 20 years the phonograph record as we know it today will be retired and in its place we will have a record which will be manufactured with a photographic process and the reproducer will scan this record electrically thus eliminating all mechanical parts. In twenty years we will count all our achievements and nostalgically look back to the "good old days."

JOHN T. MULLIN
Mincom Division, 3M Company

Studio recording techniques have changed markedly since the introduction of stereo tape recording. Today, eight-track, one-inch tape systems are providing master recording facilities of great flexibility, particularly enabling small groups of electronically oriented musicians to achieve acoustical effects never before heard in the history of music.

Being introduced at this time, are sixteen- and twenty-four-track mastering recorders, utilizing two-inch tape. Their potential for the creative recording artist will be enormous.

What of the future? We will ultimately find a practical limit to the number of tracks that may be put on a single tape. Then may come development of a continuous cross scanning system similar to that employed in video tape recorders, with the record head multiplexed at video switching rates to provide up to possibly 200 separate inputs, each sampled about 45,000 times per second. Synchronous multiplex switching on playback would permit selection and mixing of any channel or channels as desired.

A special microphone may be developed, having an extremely narrow pick up angle which may be scanned across a full symphony orchestra in synchronism with the cross scanning of the tape, resulting in a "continuum" recording wherein any small area of the orchestra may be later mixed and balanced against the others. Ten or twenty years should see solution to some relevant problems which seem impossible today. Among these are electrically scanning the narrow acoustical beam at rates up to 45 kHz and microphone response up to several megahertz. System noise must be retained at extremely low levels and the vast quantities of information must be recorded with minimum consumption of tape.

LOREN L. RYDER
Nagra Magnetic Recorders, Inc.

My field is sound for motion pictures. I can well remember back in 1929, when working at Paramount, I designed one of the first portable sound recording trucks—it weighed 9½ tons. Today I am the U. S. representative of Kudelski, and we sell the 14-pound self-powered Nagra—a weight reduction factor 1357 to 1.

During the last twenty years most consumer products have been made to give better service, and to be more trouble free, simple to operate, and more attractive. For the most part they have found a correct size and weight for utility. Look at the automobile. My present car is about the same size and weight as my 1940 car, but my present car has been 50,000 miles and it has never failed. My color television set is now over ten years old. It has had one complete replacement of all tubes and a tune up. This is reliability.

In the field of professional sound motion pictures, I do not expect much reduction in size and weight of production recording equipment. The great improvement in the future will be in the "in plant" handling of sound for better dramatic presentation. Our present compositing procedure where we combine dialogue, sound effects, and music are based on the procedure developed in the early 1930's. They are obsolete and everybody knows it—oh! what an opportunity for a break through.

The greatest change is going to take place in sound movies for the home. "Do it yourself sound movies." All the present systems are too cumbersome, too expensive and too unreliable. I venture that this change will not take twenty years. The big thrust forward should be in two to five years.

The new method of taking and presenting sound motion pictures in the home must be simple for the beginner, but for the more sophisticated it must have the capability of editing, inserting silent scenes and superimposing sound along with sound effects and music. Amateurs should be able to acquire enough experience so they can make a professional motion picture.

LAWRENCE J. SCULLY
Scully Recording Instruments Company

One of the questions most frequently asked of us at Scully is "Where will it end?" This question, from our friends in recording studios, refers to the confusion created in the industry in the past few years by the ever-increasing number of tracks required for today's recording techniques.

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.

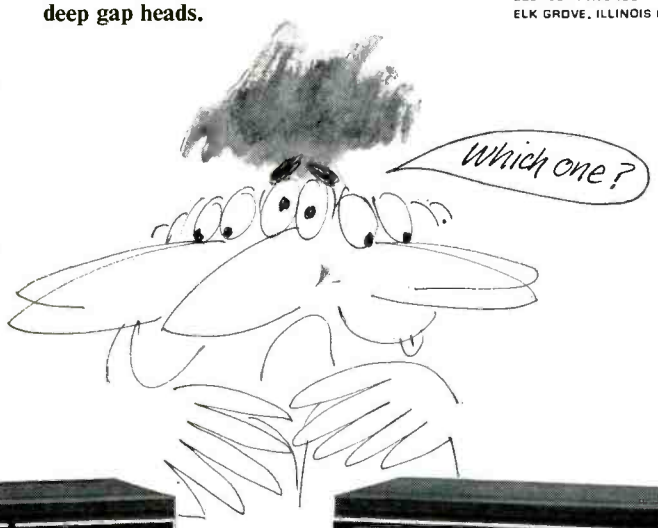
Or, pick the 1455. For lazier nitpickers, because it has automatic two-second threading and automatic reverse. Plus sound-with-sound, pause control and tape monitor. Four separate deep gap heads.

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.

AMPEX

AMPEX CORPORATION
CONSUMER EQUIPMENT DIVISION
2201 LUNT AVENUE
ELK GROVE, ILLINOIS 60007



Model 755

Model 1455

A deck for nitpickers.

And a deck for lazy nitpickers.

Check No. 35 on Reader Service Card

Prognostications

Most of us will remember when two-track stereo, on quarter-inch tape, was the only available system for multi-track recording. In the middle fifties, three-track recorders using half-inch tape became available and in the next four or five years became the standard, but not without some resistance from the "old timers" who claimed the extra channel was just a crutch to take the place of a good A&R man. With the advent of three-track recording, the multi-track proliferation was on. It became obvious that if three was good, four was better, and scarcely had three-track consoles been completed when the more progressive studios shifted over to four-track on half-inch tape. A concurrent development by Ampex Corporation at this time, Selective Synchronization, was a large factor in the popularity of the multi-track recorders. "Sel/Sync," as Ampex titled this feature, enabled the studios to re-record or over-dub on individual tracks and still be able to listen to previously recorded material on other tracks at the same time. Sync techniques became a standard feature in the recording engineer's bag of tricks and opened the way for the multi-track explosion that was to come.

Eight-track recording, on one-inch tape, was the next advance in the development, with several studios leading the way. One of the earliest eight-track studios was Atlantic Records, but the successes of some independents in the early sixties was probably most responsible for the industry move to eight-track. Our Company was to take the lead in one-inch recorders with some studios going even further and installing twelve-track units. This latter configuration was about the limit for acceptable performance on one-inch tape, but the race was on and the demand for more and more tracks continued, and the major equipment manufacturers turned to two-inch tape!

Sixteen-track recorders, on two-inch tape, are now coming into vogue and will probably become the workhorse unit in many recording studios. But, even sixteen-track isn't enough for some of the swingers! We've had requests for twenty-four and even thirty-two track recorders. However, to attempt to answer the question raised in the beginning, I feel that sixteen-track on two-inch, eight-track on one-inch, and two-track and full-track on quarter-inch will be the standard equipment in most studios in the next few years. Because of some performance limitations such as noise and cross talk, I don't expect the twelve and twenty-four configurations to become too popular, although we do expect some sales in these areas.

However, these prognostications must be considered with the knowledge that I was one of those who thought three-track and half-inch was "about as far as they can go!"

STEPHEN F. TEMMER
Gotham Audio Corporation

Audio is a strange world. It is both a business and a science, and yet the business side and the scientific side rarely if ever get together. Many business men feel that they can "make windfall profits" by not only ignoring scientific knowledge but even going contrary to it. Let us hope that the future will bring a reversal of this trend.

Undoubtedly audio's greatest need is education. There is no place in this country today that offers training or formal education to someone interested in joining this industry. That is wrong and the industry is beginning to feel the pinch. The future must bring: increased knowledge derived from newly established sources of information, increased integrity on the part of suppliers principally resulting from a more knowledgeable consumer, and genuine advancement of the art through the interchange of ideas as well as products between all of the producing and consuming nations of the world.

DANIEL R. VON RECKLINGHAUSEN
H. H. Scott, Inc.

Predicting the future is always a difficult business. If you are right, you are a prophet, and if you are wrong, you should not try predicting in the first place. However, there are certain facts and facets which are likely to materialize within the forthcoming years.

Broadcast reception will become more difficult rather than easier. The number of FM stations in the world may as much as double in the next ten years, and their radiated power will also increase. This, in turn, will make reception of distant stations more difficult because of interference from the more numerous local stations. Additionally, interference to reception from other causes will also increase. There will be more tall buildings acting as reflectors of radio waves with a consequent increase of multipath problems. To make matters worse, I believe the forest of roof-top antennas will slowly disappear for aesthetic reasons. The only way these interference problems can then be minimized will be either by the hopefully forthcoming development of a small directional indoor antenna, or by the introduction of cable FM similar to cable TV.

Other interferences will become worse. If the present trend continues, other services will be sandwiched into the present broadcast bands, as wireless microphones are at present, and the increasing use of electrical equipment will increase local interference. Again, a small antenna and the use of cable FM will help to suppress this type of interference.

What about new broadcast services? It is theoretically feasible to add a third channel to the present FM stereo broadcast system. This third channel might carry reverberation of the music being broadcast or other special effects; however, it is doubtful that such broadcasts will come into being because of the ponderous nature of the rule-making proceedings required by the Federal Communication Commission.

What about other areas? We can expect that the amplifier horsepower race will continue and that the average amplifier power will increase at the rate of approximately 5% per year. Equipment for the home is likely to remain about the same size as at present, at least as far as the front is concerned, because knobs and dials have to be operated by human beings who so far have resisted all efforts towards miniaturization. The increased power also means increased equipment dissipation, and sometime in the future the term "cool-running transistor equipment" will be forgotten because semiconductors will be developed some day which will be capable of operating at temperatures of hundreds of degrees. This is necessary to have increased power from the same size package.

There are other areas in which predictions could be attempted, such as loudspeakers, tape recorders, or records; but any such prediction would be considerably more hazardous than any of the above.

The $89.50 Miracord

with the $129.50 features

The new Miracord 620 has most of the features more expensive record changers offer plus some that are exclusive to Miracord.

For example—a tonearm that is dynamically balanced in all planes by means of an adjustable counterweight. A gram calibrated knob and pivot bearing for precise tracking force adjustment. Continuously adjustable anti-skating compensation, contoured to exactly the correct value for every point on the surface of the record. Remarkably precise cueing. The ability to track any cartridge at its recommended stylus setting to well below 1 gram. A balanced 4-pole induction motor for precise speed accuracy. A heavy pressure-formed turntable platter for smooth, steady motion. **That's what the 620 offers that other automatic turntables offer!**

Here's what other automatic turntables don't offer. Light touch pushbutton operation—a Miracord exclusive. The gentlest touch puts the Miracord into automatic play—up to 10 records. Or you can ignore the pushbuttons and play the single records manually by simply placing the arm on the record. Another Miracord exclusive lets you repeat the same record over and over. That's how easy it is to operate the Miracord 620 and to enjoy its performance.

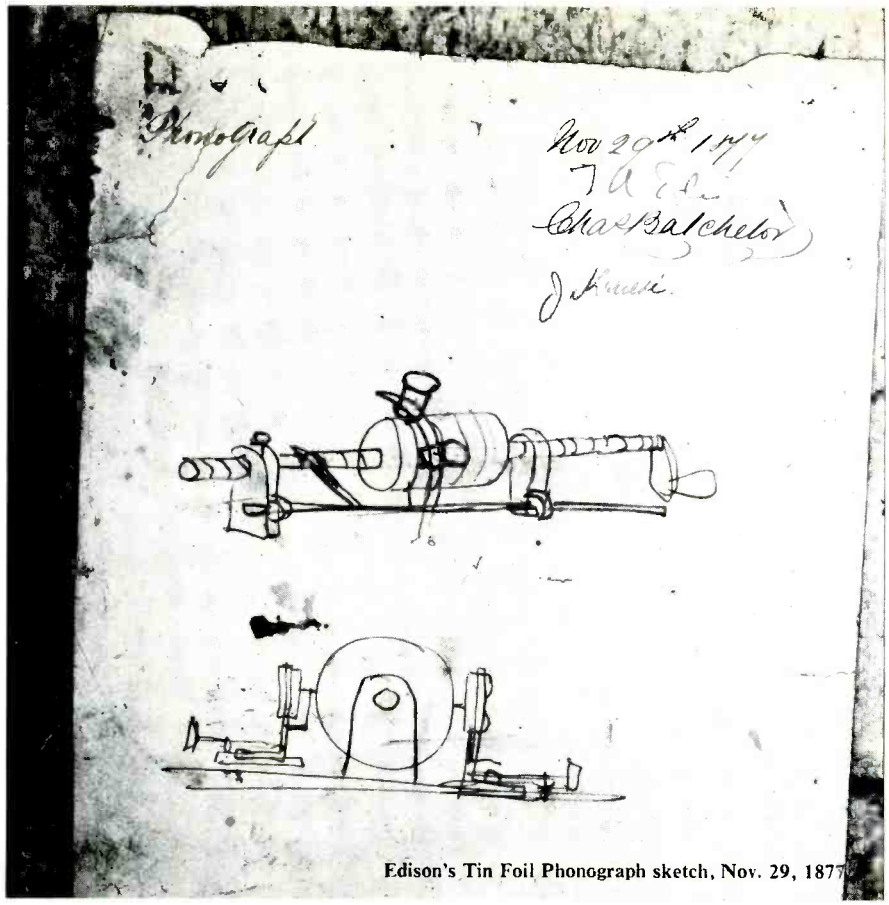
The Miracord 620 follows in the great tradition of the 50H ($149.50) and the 630 ($119.50). Model 620. Miracord quality at $89.50. See what we mean at your hi-fi dealer. Benjamin Electronic Sound Corp., Farmingdale, N.Y. 11735.

New Miracord 620

Check No. 37 on Reader Service Card

AES Exhibit Sampler

The following sampling of the new and interesting equipment to be shown at the AES Convention and Exhibit held at the Park Sheraton Hotel in New York City, October 21-24, illustrates some of the professional gear that serves to keep up the high standards of the audio profession. In addition to (or instead of) exhibit booths, many manufacturers have working demonstrations in suites elsewhere in the Hotel.



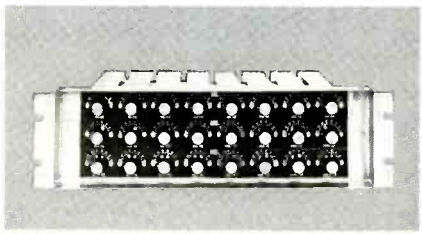
Edison's Tin Foil Phonograph sketch, Nov. 29, 1877

how it began ...and now

AKG Division of North American Philips Co. Inc. is featuring the new C-451E F.E.T. condenser microphone as a component of the Condenser Microphone Modular System. This concept offers extreme flexibility and allows the microphone to be adapted to all types of recording and sound reinforcement situations. By simply changing capsule, power source, or accessories, the user can accommodate practically any requirement. For example, there are four capsules—CK-1, cardioid; CK-2, omnidirectional; CK-6, switchable from cardioid to omni to figure-8; and CK-9, an interference tube or "shotgun" attachment. All four use the same amplifier and power supplies—either battery or a.c. These microphones use a single pair of wires with a shield, thus making it possible to use the C-451E in a system in which dynamic microphones are also used.

AKG C-451E Basic Microphone System

Altec Lansing is featuring an idea, rather than a product. The novel idea is "Acousta-Voicing," a total programmed approach to the design, installation, testing, and tuning of a high-quality sound system. The system involves complete testing of all the important parameters of the installation, including acoustic frequency response, followed by a tuning of the room in much the same way as a pipe organ is tuned to its auditorium. On completion of the "Acousta-Voicing," the system is said to be capable of driving all frequencies in the room without fear of uncontrolled feedback or reverberation which would mar the reproduction. Altec will exhibit some of the equipment used in tuning the sound installation, as well as instruct listeners in the advantages of the system.



Altec Acousta-Voice Filter Set

Ampex Corporation will display most of its professional audio line, but the "star" of the show is expected to be the new MM-1000-16—a 16-channel master recorder recently introduced. This model, like the 24-track unit, uses two-inch tape, while the similar 8-track model uses one-inch tape. Conversion from 8 to 16 to 24 channels is easily

Ampex Master Maker-1000

AKG "HUMANIZED" HEADPHONES

Humanized because . . .
research on the physiological reaction of the human ear to sound pressure led to the development of a headphone driver which functions with the human ear as a unit, and is capable of generating full fidelity sound at close proximity to the entrance of the ear.

Humanized because . . .
of their seeming weightlessness.

Humanized because . . .
of their comfortable fit which allows you to enjoy hours of listening pleasure without discomfort.

Humanized because . . .
it permits you to enjoy transparent reproduction of music and voice and still remain in partial contact with your surroundings.

TYPICAL CUSTOMER COMMENTS

"Unbelievably fantastic."

"Superior musically to six other types tested."

"Best sound yet."

"Best headphone at any price."

"Very comfortable to wear."

"Pleasure is now mine."

"Excellent sound anc comfort."

"The Best!"

"Prefer it to speakers."

"Very realistic sound."

Listen to the **AKG** K-20 or K-60 at your dealer and convince yourself.

4-68

AKG

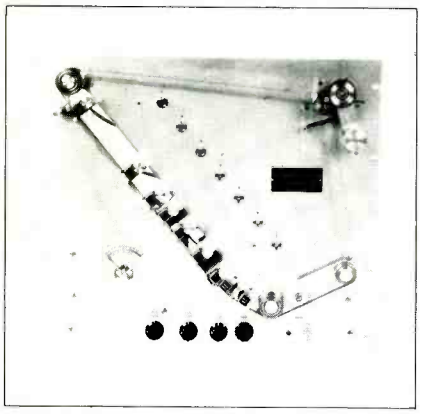
MICROPHONES • HEADPHONES

DISTRIBUTED BY
NORTH AMERICAN PHILIPS COMPANY, INC.
100 EAST 42nd STREET, NEW YORK, NEW YORK 10017

AES Exhibit Sampler

done with plug-in head assemblies, and tape guides are instantly convertible to accommodate either tape width. These recorders permit each instrument or group of instruments to be recorded separately so that balancing and blending can be done at a later date by the engineer and director after the musicians have left, allowing more time for musicians to perform at each recording session.

Audio Instrument Company, Inc. will exhibit their Model 44 continuous-loop reverberation unit. This rack mounted

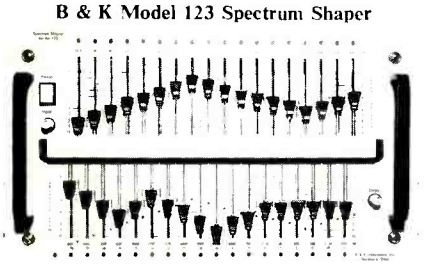


Audio Instrument Co. Tape Reverberator

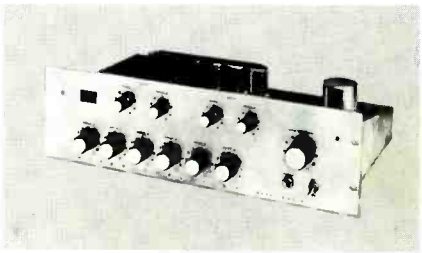
instrument utilizes a tape loop about 50 in. long which passes over an erase head, a recording head, and seven play heads. All but two of these heads may be positioned at any desired point along a slot in the panel so as to break up the repetition time to avoid the objectionable feature of tape reverberation. Separate switches permit the use of any desired number of the play heads, and control is provided for reverberation time and reverberation ratio. The entire unit is built for continuous use, and is of professional quality throughout.

B & K Instruments, Inc. will feature the Model 123 Spectrum Shaper, which permits dip-filtering of unwanted sound in precise ⅓-octave increments over the entire audio range, with each band individually adjusted from zero to more than 50 dB of attenuation. The Model 123 can be used to reject interfering sounds without noticeably affecting program material; to reject bands of noise selectively and emphasize the intelligence part of a program; to correct and equalize old records, microphones, speakers and amplifiers; and to produce special effects by selective emphasis. Vertical attenuators mounted in two rows on the front panel form a visual configuration of the spectrum shape, and a detachable memory bar recreates any previously recorded spectrum in seconds.

B & K Model 123 Spectrum Shaper



The R. T. Bozak Mfg. Co. will feature their new Model CAV-6-2 Co-ordinated Audio/Visual Perception mixer amplifier, claimed as a step forward in providing sensory perception to sound reproduction. The unit has two outputs to drive left and right speaker systems. It has six inputs, each of which is divided to drive the left and right speakers (through power amplifiers). For each input a screwdriver adjustment on the chassis permits shifting the balance between left and right outputs to co-ordinate the apparent location of

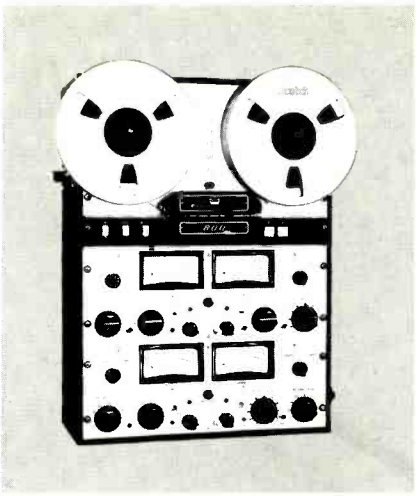


Bozak Model CAV-6-2

a sound source with the actual physical location of the microphone. Each output has is own bass and treble controls.

Capps & Co., Inc. is featuring the Capps Hot Stylus Unit which is a useful addition to the famous line of Capps styli. This unit includes a control panel and a stylus termination assembly which is easily mounted on the cutting head. The panel includes a stylus temperature control and a meter, as well as an on-off switch, pilot light, and the line cord. Models are available for most cutter heads. These units are used to apply heat to disc recording styli so as to soften the disc material at the point of contact.

Crown International will show a number of versions of their recorders using the PRO 800 computer logic control tape transport. This mechanism combines a high-quality transport with a computer which prevents damaged tapes due to inadvertent mishandling of the controls. The units are available in two- and four-channel models for



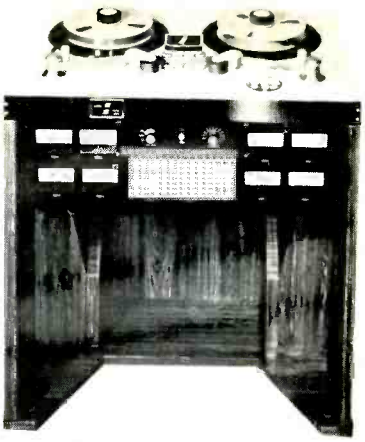
Crown 800 Series Recorder

mastering and for general studio applications.

Dolby Laboratories, Inc. will introduce a Duplicator Noise Reduction System, Model 340. A special version of the A-301 system, the new product is designed to replay noise-reduction tapes at the high speeds used in the mass production of pre-recorded tapes. With state-of-the-art techniques in high-speed duplication, the noise level on the duplicate is often limited by the build-up of noise in the several tape generations preceding the final one. The Model 340 provides a way of effectively eliminating this noise contribution. According to the manufacturer, this results in a superior quality "first generation" duplicate. Here is the typical procedure:

A record company that uses the Dolby System and wants to issue a pre-recorded tape of a particular performance can do so in two different ways. If the master recording was made with the Dolby A301, a copy of the master can be made through another A301 in the "compressed" mode. If the master tape was made by conventional means, it still can be copied through an A301, also in the "compressed" mode. The "compressed" tapes are sent to the duplication plant where a high-speed dubbing master is made with an A301, the dubbing master still being in the "compressed" mode. In the duplication process, playback signals from the high-speed tape reproducer are processed by the Model 340 and restored to normal; the signals are then fed to the high-speed slave recorders in the usual way. Since the slave recorders are receiving a signal which is the equivalent of that from the orginal master, the commercial pre-recorded tape is, in essence, a first generation

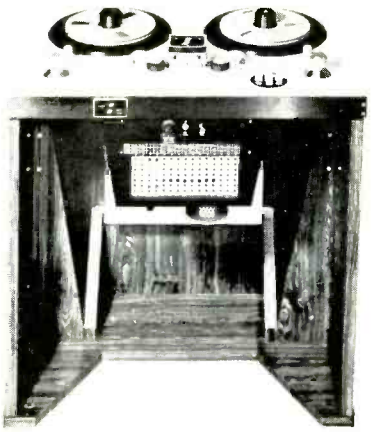
PUT THEM ALL TOGETHER . . . .



1210 Master Unit

1260 Endless Loop Bin

1220 Slave Unit



THEY SPELL ULTRA - HIGH SPEED

Combine the GAUSS 1210 Master Unit with the GAUSS 1260 Endless Loop Bin and add up to 20 GAUSS 1220 Slave units to form the most revolutionary high quality tape duplication system on the market: the new Series 1200 Ultra-High Speed Tape Duplication System. Reproduces sound with crisp, clear fidelity . . . and fast too! A 40-minute Stereo 8 cartridge program in just 18 seconds. How? The secret is FOCUSED GAP* recording. The process is uniquely GAUSS -- guarantees wider bandwidth, superior fidelity, better signal-to-noise, lower distortion, top quality. Speed? How about 240 ips -- 2 to 4 times faster than any competitor with optionally lower speeds of 60 or 120 ips. With the GAUSS 1260 Loop Bin any master tape from 10 to 1800 feet can be run continuously without stopping to recue. Time consuming rewinding is totally eliminated. Snarls and tangles JUST DON'T HAPPEN! All machine time is high quality profit-making production time. And that should be music to your ears.

*TM-GAUSS ELECTROPHYSICS, INC.

For the complete story . . . specifications, prices, delivery . . . please write or call -

GAUSS ELECTROPHYSICS, INC. an MCA INC. company

1653 12th Street, Santa Monica, California 90404 U.S.A. Telephone: (213) 451-9876 / Cable: Gauss Santa Monica

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AES Exhibit Sampler

copy. The upper frequency limit of the Model 340 is 500 kHz, which allows duplication at up to 32 times normal speed without sacrificing high-frequency response. Selection of 8, 16 or 32 times normal speed is obtained by means of interchangeable control modules. Although all modules in the system are designed for high-frequency operation, the overall complexity of the Model 340 is similar to that of the A301. Prices are reported to be the same as for the A301 system.

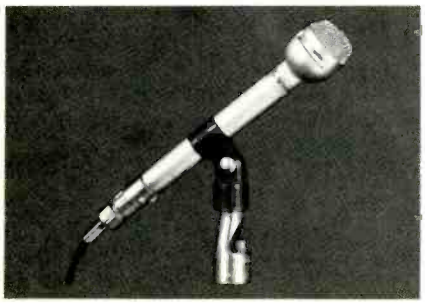
Dolby Laboratories, Inc. will continue to show the Dolby S/N Stretcher which has made such a hit in the past two shows. Capable of giving a 10-db increase in usable dynamic range plus generally cleaner, more transparent recordings on original tapes, pressings made from these master tapes are much more attractive to the listener. Already accepted by many major recording studios, the results are just now appearing in finished phonograph records, and consumers are fast becoming aware of the many advantages of the system.

Electrodyne Corporation will show samples of its elaborate studio control consoles, which incorporate all the necessary controls, switching, equalization facilities required in the preparation of master tapes in practically any conceivable studio situation.

Electrodyne ACC-1608 Console

Electro-Voice, Inc. will be showing their complete line of professional microphones, ranging in size from the 649B lavalier up to the long-range cardioid line unit Model 643. Among most recently introduced models will be the RE-15, a modestly priced unit which is rugged and yet of broadcast quality.

Electro-Voice RE-15 Microphone

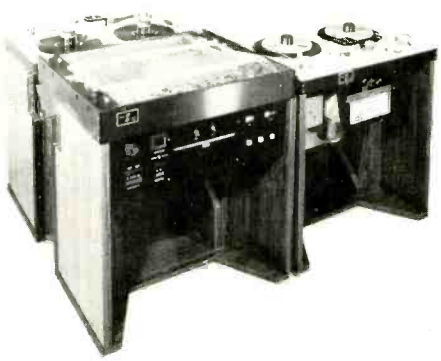


Gately Electronics will show their six-channel "console in an attache case" which makes it possible to record in locations where a mobile unit could not possibly reach. Although operating on 117 V, 60 Hz, it requires only 10 watts of power, yet provides for six inputs, all of which can be switched to either or both output channels. Output level is +24 dBm, at less than 1% THD from 20 to 15,000 Hz. Available with a variety of accessories to permit phono inputs, or mic inputs of especially low level, when required. Attache case is only 18 x 12¼ x 5¼ in. high, and is available in either black or olive finish.

Gately PM-1 Portable Mixer

Gauss Electrophysics, Inc. will be showing the new Gauss Series 1200 Ultra High Speed Tape Duplicating system which features tape speeds of 240, 120, or 60 ips with ratio of 32:1 or 16:1 duplication. The system uses solid-state electronics throughout, together with the focused-gap recording process. The revolutionary new Model 1260 horizontal endless tape bin for tape masters from 10 to 1800 ft. in length operates at speeds of 240, 120, or 60 ips. The basic system is capable of producing over 1800 8-track stereo cartridges in each 8-hour shift. As many as 20 slave recorders can be used with one master and bin to produce over 36,000 cartridges per shift.

Gauss Electrophysics Horizontal Tape Bin with Master and Slave Recorders

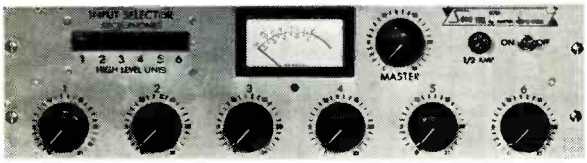


Gotham Audio Corporation will feature the complete Neumann Disc Transfer System—shown for the first time in this country. This system includes the SP-66 Transfer Console, with a Telefunken M-10 A Professional Tape Machine; a Neumann VMS-66 Computer Controlled Disc Mastering Lathe; the Neumann SX-68 Stereo Cutterhead; and the new VG-66S solid-state stereo electronics package for the SX-68. The latter is the first all-solid-state cutterhead electronics on the market. This package contains all the necessary amplifiers, equalizers, and power supply equipment for use with feedback stereo cutterheads.

A. Gotham's Neumann VMS-66 Disc-Cutting Lathe
B. Gotham's Neumann SP66S Control Console

Hewlett-Packard will feature a new real-time audio spectrum analyzer, in addition to their other acoustic measuring instruments (Sound Level Meter, Loudness Analyzer). The Audio Spectrum Analyzer, Model 8054A, looks at 24 ⅓-octave frequency bands simultaneously and writes out a spectrum in less than 30 ms, doing so at rates up to 35 spectra per second. In addition to having visual and analog outputs, the new analyzer also has digital outputs, giving it unprecedented flexibility and speed in the processing of data. The unit works this way: there are 24 filters connected in parallel at the input, and each filter has a detector that determines the amplitude of signal components in that channel. An elec-

Introducing three new products from Martin Audio



SONOMIX 601A — A NEW IDEA IN MIXERS . . . SIX CHANNELS IN EUILDING BLOCK UNITS

BAS C UNIT — Line level for tape mixing

SEC ɔND UNIT — Microphone preamp unit for live recording

THIF D UNIT — Nickel cadmium battery pack for remote recording

FOU RTH UNIT — Portable case

- 95 DB gain
- 200 Ω microphone input
- 100K Ω line input
- ±4VU 600 Ω output
- XLR connectors on microphones and line output
- 19 lbs. with case and battery pack

$675.00 COMPLETE

VARISPEED III — TAPE RECORDER EFFECTS GENERATOR

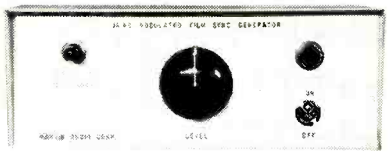
Vari:peed will vary the motor frequency of a studio recorder from 25 to 80 cycles and synchronize at 30, 48, 50 and 60 cycles.

Plugs directly into Ampex models 300-350-440, Scully models 280-282 and adapts to most other studio recorders.

INVALUABLE FOR

- Sound effects
- Correcting off speed tapes
- Variable sound delay and changing pitch
- Playing 3¾ IPS tapes on 7½ IPS machines
- Operating 60 cycle recorders on 50 cycles
- Synchronizing film tracks

$395.00



STUDIO SYNC GENERATOR FOR 14000 CYCLE MODULATED FILM SYNC

Solid State modulated signal generator for recording motion picture sync tracks on ¼ inch and multi-track non sprocket tape recorders.

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Telephone 212-661-6987

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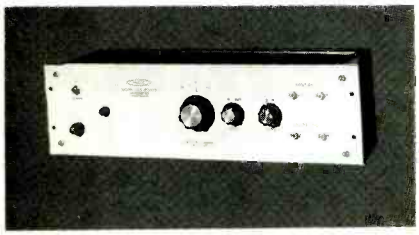
AES Exhibit Sampler

Hewlett-Packard Audio Spectrum Analyzer

tronic switch sequentially samples the detector outputs in sync with a CRT sweep to display amplitude vs. frequency. The speed of scanning (1 ms per channel) presents the data virtually in real time.

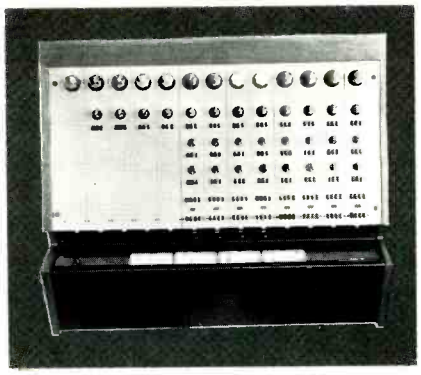
Holzer Audio Eng. Co. (HAECO) is showing the new "esg"—compatable stereo generator—a device which makes it possible to combine the two channels of a stereo program into a monophonic signal without doubling up



HAECO Compatible Stereo Generator

the "center," as is done when the two channels are simply combined by paralleling. This unit thus makes it possible for an AM station, for example, to broadcast stereo recordings monophonically with a program balance more closely following what would be obtained if it had been recorded originally in the mono mode. As a matter of fact, the re-creation can be made to have less than the correct amount of "center" if such a condition should be required. The encoding of the signal will have no significant effect on the stereo, but the mono reproduction will be definitely enhanced, according to the manufacturer.

Langevin will be featuring their AM4A mixer assembly, which offers a new concept in a sound mixer. The building-block concept lets you buy only the channels you need and drop them into the AM4A housing, which has an overall height of 7 in. and a depth of 27 in. Width will vary with the number of modules installed. The sys-



Langevin Mixer Console

tem basically includes the required number of AM401 input modules, each of which provides amplification, gain control, equalization, and switching—both program and reverb lines. The AM407 Program Module receives the outputs of the various input modules and provides for "reverberation point" selection so that reverberation can be recorded or just monitored without being recorded. The MG61 master gain control provides for control of the entire system, as required.

Metrotech, Inc. will display its new line of professional recorders. The 500A Series includes straight-line threading, automatic tape lifters, inter-

Metrotech 500A Series Recorder

locked controls, edit/cue functions, and two-speed motors with automatic equalization switching. Operating convenience is one of the principal advantages of the new design. A hinged head cover provides easy access for tape editing.

The R. A. Moog Company again brings its wide line of electronic music composing equipment to the attention of the engineer. These synthesizers are integrated systems of modular, single-frequency generating, modifying, and controlling instruments, all designed to perform their functions over a wide range, and with an accuracy consistent with the state of the art. The use of these instruments with one or more tape recorders offers a great flexibility to composers in this new medium of musical expression.

Nagra Magnetic Recorders, Inc. and Ryder Magnetic Sales Corp. will be featuring the Nagra portable tape recorder. Available in both synchronous and non-sync models, the Nagra is normally equipped with a mixing corcuit for one microphone and a line or bridging circuit, but with the addition of another microphone preamp, two microphones can be mixed readily. The unit works at 15, 7½, and 3¾ ips. and will accommodate 5-in reels with the cover closed, 7-in reels with the cover

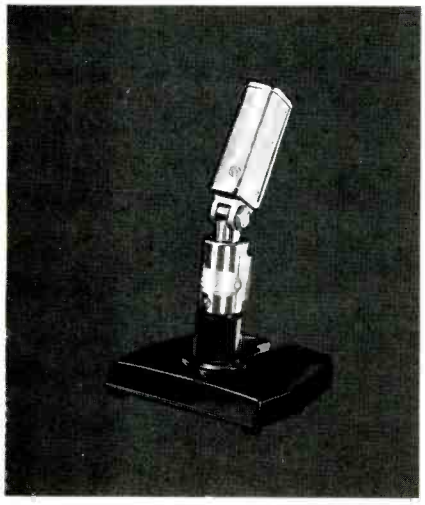
Nagra Portable Sync Recorder

open, and works from a supply consisting of 12 standard "D" cells. The Nagra measures 12½ x 8.7 x 4.3 in., and weighs just under 14 lbs. Speed regulation is claimed to be .05 per cent or better, and frequency response at 15 ips is from 30 to 18,000 Hz ±1 dB, and at 7½ ips it is from 40 to 15,000 Hz ±1 dB.

Sennheiser Electronic Corporation will feature their line of high-quality condenser microphones designed for studio and professional applications. Among them will be the MKH-110 which is a special low-frequency test unit, and the MKH-110/1, similarly good on the extremely low frequency ranges but capable of accepting high sound pressures up to 5000 microbars. Two lavalier models—both condenser types—will be shown, Models MKH-124 and MKH-125. In addition, test instruments and headphones will be exhibited.

Shure Brothers, Inc. will exhibit their line of studio dynamic microphones in both cardioid and omnidirectional

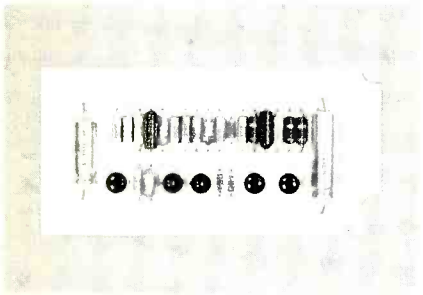
AES Exhibit Sampler



Shure Brothers Model 330 Ribbon Microphone

types, together with their line of microphone accessories. Shure's professional mixer and its Audio Level Controller are also expected to be displayed.

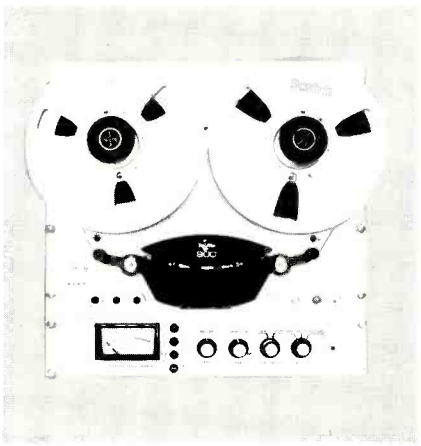
Spectra-Sonics will show the Model 101 Audio Amplifier, a solid-state circuit card offering a gain of 40 dB and an output of +18 dBm. The low noise level permits its use in any application in an audio control system from micro-



Spectra-Sonic Model 101 Audio Amplifier

phone preamp to line output amplifier. The amplifier performs as if terminated by an external isolation transformer—thus providing freedom from circuit grounds without the size or expense of transformers. Source impedance ranges from 50 ohms to ∞ and output loading can range from 600 ohms to ∞, and it is stable under any condition of pure capacitive loading at either input or output.

Tape-Athon Corp. will show the new Model 900 Logger, intended for application requiring long-time recording of broadcast material. It operates at $1\frac{7}{16}$ or $1\frac{5}{32}$ ips, with wow and flutter specifications of less than 0.3 and 0.5 per cent, respectively, at the two speeds. The Logger is equipped with an automatic reversing feature at each end of reels to assure continuous recording.

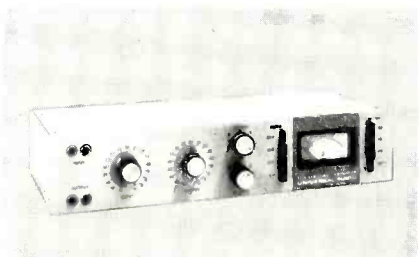


Tape-Athon Model 900 Logger

Using triple-play tape, the 900 will log over 400 hours of broadcasting on a single $10\frac{1}{2}$-in. reel. Frequency response at $1\frac{5}{32}$ ips is claimed to be from 200 to 3000 Hz with a S/N of 38 dB, unweighted.

The 3M Company introduces a compact, remote overdub control as a professional mastering accessory. The small control, which measures 6 by 6 by 4 inches, fits easily into recording consoles. In addition to remotely and electronically controlling overdubbing, it also helps eliminate the possibility of accidental erasure. The control is available as a factory-installed accessory for 3M professional recorders at a cost of $950.

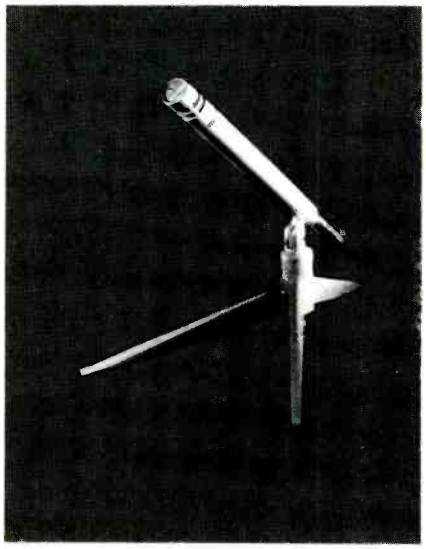
United Recording Electronics Industries is presenting the Universal Audio 1176 Solid-State Limiting Amplifier which utilizes an F.E.T. as a voltage-variable resistor ahead of the first stage of amplification. Unique circuitry permits severe limiting without added distortion, and no balancing is ever required,



Universal Audio 1176 Limiting Amplifier

says U.R.E.I. Attack time is adjustable from less than 20 microseconds to 800 microseconds. This means that a 50-kHz peak is fully stabilized at the limited level within one cycle. Release time is also adjustable from the front panel from 50 ms to 1.1 seconds. The 1176 is rack mounted, and requires only $3\frac{1}{2}$ in. of vertical space. Self-contained regulated power supplies operate from either 110-125 or 220-240 V a.c., 50 or 60 Hz.

Vega Electronics Corp. will be featuring the Vega S-10 Condenser Microphone, a solid-state unit with self-contained battery. This is a cardioid model, finished in beige, with outputs



Vega S-10 Condenser Microphone

of 50 ohms and high impedance. Output coupling is by means of a Cannon plug, and a 20-foot cable is supplied. Also available is the S-10B, which is the same except for an omnidirectional pattern. A.c. supplies are available for both models.

Wiegand Audio Laboratories will be exhibiting their new line of modular mixing console components and complete factory-wired systems. The W.A.L. Model 300 Master Record-Reverb Module provides a master record straight-line attenuator, two 22-input active combining networks, a built-in solid-state reverb system, external reverb line input and output, direct line output, a master echo-send edgewise-mounted VU meter, together with a master echo-send control and two echo-return-level mixers. In addition, a built-in reed relay is employed to place slating information on the tracks in the record mode. Another attraction of the W.A.L. booth will be a Scully model 284-8 eight-track recorder operating with the W.A.L. console system.

Wiegand Model 300 Record-Reverb Module

Make Christmas A Heathkit® Holiday

Wish Your Family Merry Christmas This Year With A New Heathkit Color TV . . . A Better Buy Than Ever With New Lower Prices

NEW Deluxe Color TV With Automatic Fine-Tuning—Model GR-681

kit GR-681 **$499.95** (less cabinet)

The new Heathkit GR-681 is the most advanced color TV on the market. A strong claim, but easy to prove. Compare the "681" against every other TV — there isn't one available for any price that has all these features. Automatic Fine Tuning on all 83 channels . . . just push a button and the factory assembled solid-state circuit takes over to automatically tune the best color picture in the industry. Push another front-panel button and the VHF channel selector rotates until you reach the desired station, automatically. Built-in cable-type remote control that allows you to turn the "681" on and off and change VHF channels without moving from your chair. Or add the optional GRA-681-6 Wireless Remote Control described below. A bridge-type low voltage power supply for superior regulation; high & low AC taps are provided to insure that the picture transmitted exactly fits the "681" screen. Automatic degaussing, 2-speed transistor UHF tuner, hi-fi sound output, two VHF antenna inputs . . . plus the built-in self-servicing aids that are standard on all Heathkit color TV's but can't be bought on any other set for any price . . . plus all the features of the famous "295" below. Compare the "681" against the others . . . and be convinced.

GRA-295-4, Mediterranean cabinet shown **$119.50**
Other cabinets from $62.95

Deluxe "295" Color TV...Model GR-295

now only **$449.95** (less cabinet)

Big, Bold, Beautiful . . . and packed with features. Top quality American brand color tube with 295 sq. in. viewing area . . . new improved phosphors and low voltage supply with boosted B+ for brighter, livelier color . . . automatic degaussing . . . exclusive Heath Magna-Shield . . . Automatic Color Control & Automatic Gain Control for color purity, and flutter-free pictures under all conditions . . . preassembled IF strip with 3 stages instead of the usual two . . . deluxe VHF tuner with "memory" fine tuning . . . three-way installation — wall, custom or any of the beautiful Heath factory assembled cabinets. Add to that the unique Heathkit self-servicing features like the built-in dot generator and full color photos in the comprehensive manual that let you set-up, converge and maintain the best color picture at all times, and can save you up to $200 over the life of your set in service calls. For the best color picture around, order your "295" now.

GRA-295-1, Walnut cabinet shown **$62.95**
Other cabinets from $99.95

Deluxe "227" Color TV...Model GR-227

now only **$399.95** (less cabinet)

Has same high performance features and built-in servicing facilities as the GR-295, except for 227 sq. inch viewing area. The vertical swing-out chassis makes for fast, easy servicing and installation. The dynamic convergence control board can be placed so that it is easily accessible anytime you wish to "touch-up" the picture.

GRA-227-1, Walnut cabinet shown **$59.95**
Mediterranean style also available at $99.50

Deluxe "180" Color TV...Model GR-180

now only **$349.95** (less cabinet)

Same high performance features and exclusive self-servicing facilities as the GR-295 except for 180 sq. inch viewing area. Feature for feature the Heathkit "180" is your best buy in deluxe color TV viewing . . . tubes alone list for over $245. For extra savings, extra beauty and convenience, add the table model cabinet and mobile cart.

GRS-180-5, table model cabinet and cart **$39.95**
Other cabinets from $24.95

Now, Wireless Remote Control For Heathkit Color TV's

Control your Heathkit Color TV from your easy chair, turn it on and off, change VHF channels, volume, color and tint, all by sonic remote control. No cables cluttering the room . . . the handheld transmitter is all electronic, powered by a small 9 v. battery, housed in a small, smartly styled beige plastic case. The receiver contains an integrated circuit and a meter for adjustment ease. Installation is easy even in older Heathkit color TV's thanks to circuit board wiring harness construction. For greater TV enjoyment, order yours now.

kit GRA-681-6, 7 lbs., for Heathkit GR-681 Color TV's **$59.95**
kit GRA-295-6, 9 lbs., for Heathkit GR-295 & GR-25 TV's **$69.95**
kit GRA-227-6, 9 lbs., for Heathkit GR-227 & GR-180 TV's **$69.95**

Now There Are 4 Heathkit Color TV's . . . All With 2-Year Picture Tube Warranty

kit GR-681

kit GR-295

kit GR-227

kit GR-180

New Wireless TV Remote Control
For GR-295, GR-227 & GR-180
$69.95

New Wireless TV Remote Control
For GR-681
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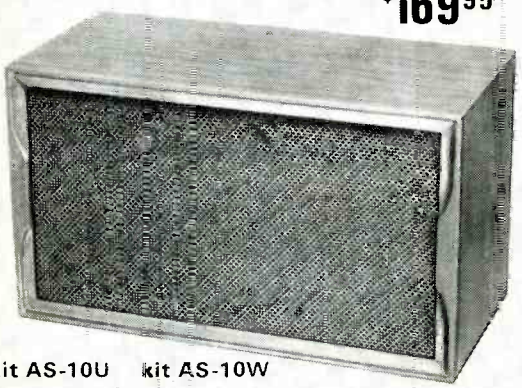
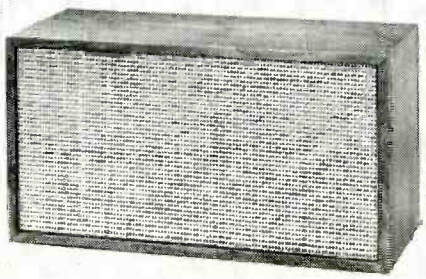
NOW, THE TUNER AND AMPLIFIER OF THE HEATH AR-15 RECEIVER ARE AVAILABLE AS SEPARATE COMPONENTS

NEW kit AJ-15 $189.95*

NEW kit AA-15 $169.95*



kit AS-10U $59.95 (unfinished) kit AS-10W $64.95 (walnut)



kit AS-16 $49.95

HEATHKIT AR-15 Deluxe Stereo Receiver

The World's Finest Stereo Receiver . . . the Heathkit AR-15 has received high praise from every leading audio & electronics magazine and every major consumer testing organization. Here are some of the many reasons why. The AR-15 delivers 150 watts music power from its 69 transistor, 43 diode, 2 IC's circuit — 75 watts per channel. Harmonic and IM distortion are both less than 0.5% at full output for clean, natural sound throughout the entire audio range at any listening level. The FM tuner has a cascode 2-stage FET RF amplifier and an FET mixer to provide high overload capability, excellent cross modulation and image rejection. The use of crystal filters in the IF section is a Heath first in the industry and provides an ideally shaped bandpass and adjacent channel selectivity impossible with conventional methods. Two Integrated Circuits in the IF amplifier provide hard limiting, excellent temperature stability and increased reliability. Each IC is no larger than a tiny transistor, yet each contains 28 actual parts. The FM tuner boasts sensitivity of 1.8 uV, selectivity of 70 dB and harmonic & IM distortion both less than 0.5% . . . you'll hear stations you didn't even know existed, and the elaborate noise-operated squelch, adjustable phase control, sterel threshold control and FM stereo noise filter will let you hear them in the clearest, most natural way possible. Other features include two front panel stereo headphone jacks, positive circuit protection, transformerless outputs, loudness switch, stereo only switch, front panel input level controls, recessed outputs, two external FM antenna connectors and one for AM, Tone Flat control, a massive electronically filtered power supply and "Black Magic" panel lighting. Seven circuit boards & three wiring harness make assembly easier and you can mount your completed AR-15 in a wall, your own custom cabinet or the rich walnut Heath cabinet. For the finest stereo receiver anywhere, order your AR-15 now. 34 lbs. *Optional walnut cabinet AE-16, $24.95.

HEATHKIT AJ-15 Deluxe Stereo Tuner

For the man who already owns a fine stereo amplifier, Heath now offers the superb FM stereo tuner section of the AR-15 receiver as a separate unit. The new AJ-15 FM Stereo Tuner has the exclusive FET FM tuner for remarkable sensitivity, exclusive Crystal Filters in the IF strip for perfect response curve and no alignment; Integrated Circuits in the IF for high gain, best limiting; Noise-Operated Squelch; Stereo-Threshold Switch; Stereo-Only Switch; Adjustable Multiplex Phase, two Tuning Meters; two Stereo Phone jacks; "Black Magic" panel lighting. 18 lbs. *Walnut cabinet AE-18, $19.95.

HEATHKIT AA-15 Deluxe Stereo Amplifier

For the man who already owns a fine stereo tuner, Heath now offers the famous amplifier section of the AR-15 receiver separately. The new AA-15 Stereo Amplifier has the same superb features: 150 watts Music Power; Ultra-Low Harmonic & IM Distortion (less than 0.5% at full output); Ultra-Wide Frequency Response (±1 dB, 8 to 40,000 Hz at 1 watt); Front Panel Input Level Controls; Transformerless Amplifier; Capacitor Coupled Outputs; All-Silicon Transistor Circuit; Positive Circuit Protection. 26 lbs. *Walnut cabinet AE-18, $19.95.

HEATHKIT AS-10 Acoustic Suspension System

This high performance Heathkit system features the extended bass response, smooth high frequency response and low distortion that have made acoustic suspension systems a favorite of thousands. The 10" woofer produces rich bass down to 30 Hz, and the two 3½" tweeters deliver clean, natural highs up to 15 kHz. The high frequency level control on the back of the factory assembled cabinet lets you adjust the sound the way you like it and the system requires only 10 watts to drive it, yet handles up to 40 watts of program material. Easy, enjoyable one evening assembly — just wire the 2250 Hz L-C type crossover, mount the speakers and sit back and enjoy the amazing performance. The rich walnut of the assembled cabinet goes with any decor, or order the AS-10U and have the added pleasure of putting the finish of your choice on it. Install either horizontally or vertically. Order two for superb stereo now. 43 lbs.

HEATHKIT AS-16 Compact 2-Way System

Don't let the small size and low cost fool you — the AS-16 performs with an authenticity comparable to many higher priced, larger systems. The 8" acoustic suspension woofer and two 3½" tweeters have smooth, lifelike response from 45 Hz to 20,000 Hz, without distortion or unnatural emphasis. The high frequency level control lets you balance the highs to suit your taste. Handles from 10 to 25 watts of program material and the compact 10" H x 19" W x 8¼" D walnut veneer cabinet is covered with a clear, tough vinyl to protect against spills and scratches. Goes together in just 2 hours . . . the speakers are already cabinet-mounted — just wire the 1500 Hz crossover network. Buy two of these excellent Heathkit systems now and enjoy remarkable stereo at a reasonable cost. 22 lbs.

Gift For Everyone On Your List

HEATHKIT AD-27 FM Stereo Compact

The new Heathkit AD-27 produces stereo sound comparable to many very good stereo systems, for the simple reason that it wasn't engineered to meet the usual performance standards of compacts. Heath engineers took their top rated AR-14 solid-state stereo receiver, modified it physically to fit the cabinet, and matched it with the excellent British-crafted BSR McDonald 500A Automatic Turntable. The result is the Heathkit "27" Component Compact. Here it is in detail: The amplifier delivers an honest 15 watts music power per channel — enough to drive any reasonably efficient speaker system . . . ±1 dB response from 12 Hz to 60 kHz . . . channel separation is a remarkable 45 dB. Harmonic & IM distortion are both less than 1% at full output. The advanced transformerless output circuitry provides lower phase shift and lower distortion plus protection against transistor damage from shorted output leads. The performance of the FM Stereo tuner section is nothing short of outstanding. A flip of the rocker-type power switch and the 31 transistor, 10 diode circuit is ready to go. Tune across the dial with the smooth inertia flywheel tuning . . . the clarity & separation will amaze you and you'll wonder where all those stations were before. Poor separation is eliminated thanks to the adjustable phase control and AFC puts an end to drift. Stereo indicator light, filtered tape outputs and a low noise electronically filtered power supply too. The precision BSR McDonald automatic turntable has features normally found only in very expensive units, like cueing and pause control, variable anti-skate device, adjustable stylus pressure, low mass tubular aluminum tone arm with a famous Shure diamond stylus magnetic cartridge and automatic system power too — the turntable will turn the system on & off. The beautiful walnut cabinet with sliding tambour door will be a welcome addition to any room too. For the finest stereo compact on the market, get your "27" Component Compact now. 41 lbs.

NEW kit AD-27
$169.95

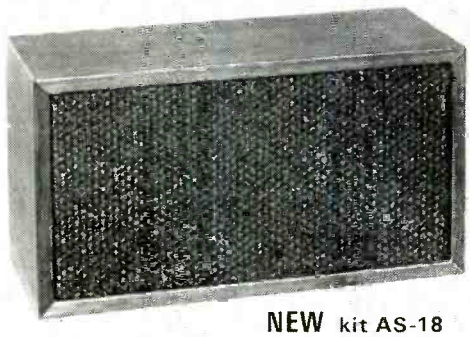
HEATHKIT AD-17 Low Cost Stereo Compact

This new Heathkit Stereo Compact delivers quality stereo sound at a budget-saving price. By taking the stereo amplifier section of the AD-27 above and combining it with the top performing BSR McDonald 400 Automatic Turntable, Heath engineers were able to put together a stereo package that outperforms anything in its price class by a wide margin. And here's the AD-17 close-up. The 17 transistor, 6 diode amplifier puts out a husky 15 watts music power per channel — sufficient power to drive most speaker systems. Harmonic & IM distortion are both markedly less than other compacts in this range — less than 1% at full output. Channel separation is 45 dB. Front panel dual-tandem controls for Volume, Bass and Treble let you adjust the sound to your liking and the variable Balance control eliminates annoying level differences between right and left channels. A stereo headphone jack is conveniently located near the recessed inputs on the side of the cabinet. A front panel speaker on-off switch lets you turn off the speakers for private headphone listening. Tuner and auxiliary inputs allow you to add the enjoyment of FM stereo and tape recording later if you wish. The high quality BSR McDonald 400 Automatic Turntable features a variable cueing and pause control, adjustable stylus pressure adjust, adjustable anti-skating and many more precision features normally associated with turntables costing much more. Comes equipped with a famous Shure magnetic cartridge too. Easy, enjoyable 12-15 hour assembly is assured through the use of circuit board, wiring harness construction and the easy to understand Heathkit manual. Just wire the circuit board and install the assembled turntable in the handsome walnut finish cabinet . . . you'll have a stereo compact that will look nice and perform great — the Heathkit AD-17. Order yours today. 28 lbs.

NEW kit AD-17
$109.95

HEATHKIT AS-18 Miniature Speaker System

The new Heathkit AS-18 will remove your suspicions about the performance of miniature speaker systems forever. Physically it's only 8¼" H x 15¼" W x 6½" D but it will outperform many larger systems that cost much more. Heath engineers used well-known high quality Electro-Voice® speakers and good design methods to produce the most surprising little speaker system you've ever heard. The 6" acoustic suspension woofer produces full, rich bass down to an 60 Hz and the 2½" tweeter delivers clear, natural highs up to 20 kHz — excellent performance for most any system. A high frequency balance control lets you adjust the sound to suit you. Handles 25 watts of program material. The speakers mount from the front of the clear vinyl covered cabinet for easier assembly and better sound. The AS-18 makes an ideal performance companion to either of the new Heathkit Component Compacts above, and its perfect for anywhere you need superior performance from a small space. Pick up a pair of these startling little performers for stereo. 16 lbs.



NEW kit AS-18
$32.95

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FREE 1969 CATALOG!
Now with more kits, more color. Fully describes these along with over 300 kits for stereo/hi-fi, color TV, electronic organs, electric guitar & amplifier, amateur radio, marine, educational, CB, home & hobby. Mail coupon or write Heath Company, Benton Harbor, Michigan 49022.

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The Art of the Small Recording Studio

JOSEPH GIOVANELLI

THERE ARE MANY recording studios whose products will never make the "Top Ten," nor are they intended to. This article will touch upon the activities of these studios, their probable equipment requirements, and the problems they face.

How small is small? This covers a wide range, and the answer about size will provide an idea of the type of work in which such studios are engaged. A "studio" might merely consist of a music teacher with a tape recorder who tapes and sells to his students copies of their performances. It may be someone who occasionally records a wedding ceremony with a view to selling the recording to the bride and groom. The small recording company might be run by one man or a group of men. Perhaps they make recordings for would-be actors or singers to use as audition material, or for teachers to use as educational material. Probably the small recording company will do all of these things and more besides.

Generally speaking, all small recording studios have one thing in common. Theirs is a custom service. They do *not* produce records for the mass market. They may make no more than one or two copies from a given taping session. Thus, each tape produced is custom made to suit the individual requirements of the client. Whether it is rock or chamber music, the products are sold to the performers, seldom to the general public.

The recordist might go to a high-school spring concert, tape it, and sell tapes or discs to the performers and to the audience. It would be impossible for the large record companies with their high overhead in terms of manpower, plant, and equipment to record this same high-school concert and sell the products at a price the audience could afford to pay. It is obvious that the small studio is not competing with the giants for the same market. Therefore, their requirements in terms of equipment, personnel, and studio facilities are considerably different.

The first consideration is the studio. Some recording companies may not even have a studio, especially those who specialize in "dubbing" or making copies from previously recorded material onto tapes or discs. This specialty might be narrowed even further to "air checks," the recording of programs from radio or television in the hope of selling their product to the participants on such shows.

However, most small recording companies do have some kind of studio. Such studios are usually small because the number of people being recorded at any given time rarely exceeds half a dozen or so. This does not mean that the small companies do not record large groups of performers, however. In such cases they may rent a hall for the purpose or use the group's own rehearsal facilities as the recording studio. Of course, there are studios with many rooms where several recording sessions or rehearsals can go on at the same time, but these still come under the classification of a small studio as opposed to the major recording companies.

Equipment Needs

The amount of recording and auxiliary equipment that the small studio may own will vary, depending on the type of work it performs, its budget, and the ingenuity of its personnel. A company that specializes in dubbing obviously needs no microphones and no console. All that is required is sufficient tape or disc equipment to carry out the necessary functions. They may or may not require some kind of equalizer to compensate for deficiencies in clients' tapes. If the work involves recordings from radio or television, they will often have the best obtainable AM and FM receivers, and the best possible equipment for recovering sound from the television channels. This last is not easily obtainable commercially. It is often specially built by or for the studio.

When the company offers studio facilities—and most of us do—some kind of mixing equipment will be required. Generally, budget limitations prevent ownership of eight- or twelve-track equipment. We must make do with two-channel stereo equipment, using these two channels in much the same way that large recording companies use eight or twelve tracks. Over-dubbing must be accomplished with a second recorder rather than by taping track upon track, with a view to later re-mixing.

The lack of such multi-track equipment imposes some considerable difficulties on the engineer doing the mixing. If he does a poor mix, he has no recourse but to re-record a given track and hope that his mix on this particular over-dub was better than his previous mix. This calls for a considerable amount of artistry, and most of us enjoy the challenge. After all, excellent work was turned out before the advent of all this multi-track equipment. Therefore, small studios are not doing the impossible in terms of producing a fine product with limited equipment—limited in terms of what is available to the larger companies. In truth, much of recording is art, not science. It is the ability of the engineer which makes the difference.

While not having the flexibility of the larger companies, one can, with proper care and patience, put out a product which is as good as, and very often better than, a similar product marketed by a large record company. Part of the reason for this is being able to give a particular session more attention than it can be accorded in a large company whose schedule is tight.

Another reason is that the number of different pieces of equipment used is far fewer than will be found in the chain used by a large studio. For example, if able to make a successful recording using a two-track original, this master tape is used to make the final discs or tapes for the client. This is seldom possible in large installations where virtually all taping is multi-track, requiring a mix-down before duplication to the master disc. In addition to the fact that master tapes are very often the originals, the number of amplifiers, equalizers, limiters, and other recorders through which the signal must pass on its way from the microphone to the final product is usually far fewer in number in a small studio than would be true of the large house. Therefore, one can end up with less distortion and with a wider frequency range.

One piece of equipment which is seldom involved in a re-mixing process in a small studio, but certainly is in a large one, is a recording console. I believe that a console for the small studio is unnecessary since we are not recording more than two simultaneous channels. Of course, each channel may be fed from a mix consisting of several inputs. A mixer will serve the purpose as well or perhaps better than a console. True, small studios lack the refinements of separate equalizers and reverb facilities on each input, but for their purposes these features are seldom necessary. Superb results can be obtained by using overall echo, which

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can be introduced at any time during the overdubbing process.

Microphones must be given careful consideration. Unlike the larger studios, the small studio, with a small budget, must work with as few microphones as possible. Certainly we need two high-quality condenser microphones. Generally these will be of the omnidirectional type.

Then, too, small studios are often plagued with poor acoustic conditions when making remote recordings. Omnidirectional microphones will only aggravate this situation. Therefore, I believe that the small studio needs at least two microphones whose low-frequency range is limited, and whose pickup pattern is both cardioid and close-talk. Such a microphone will enable a singer to work as close to it as he wishes in order to eliminate pickup from other instruments or to eliminate room reverberations. The lack of low-frequency response from such a microphone makes it unnecessary to employ separate equalization on that mixer input—an equalization capability which we might not even have.

If possible, the studio should also probably have a pair of cardioid microphones which have a wide frequency range, in addition to the close-talk mike just described. Thus, with half-a-dozen mikes we are in a position to meet the demands of most acoustical environments in which small-studio recordists are likely to find themselves.

Remote recordings also pose problems since, for the most part, equipment specifically for location recordings is generally not available. At least one of the recorders and the mixer must be readily detachable from the remainder of the control room equipment so that it can serve as the remote recording equipment.

Up to this point we have only discussed the equipment required to make the original recording. If the customer wishes a tape copy of the session, it is no problem to copy from the master tape as often as desired, but many clients will require disc copies. This requires disc recording equipment. Discs are far more difficult to make than tapes.

Some small studios own their own disc recording equipment, but others prefer to make their own tapes and farm out their disc recording work. Unless the operator of a small studio is fascinated with the problems of disc recording, farming out his disc work is a very logical approach to the problem. Also, unless he is mechanically inclined and can build his own, a good disc recording lathe will cost him several thousand dollars. In addition, the recording head will cost a considerable sum, especially if he plans to make stereo discs.

In any case, if a person has never cut a disc before, it will take him several months of frustration before he learns to do so successfully. When you stop to consider the investment in equipment and time, the farming out of your disc recording work looks very attractive. I have made discs for more than twenty years, so I know just what a novice to this art must go through. His problems are made more difficult as the state of the art places more and more exacting demands on the disc maker.

We now have traced the activities of the operator of a small studio through the type of work that is performed and the equipment used to perform that work. The ability to make high-quality tapes and discs is meaningless unless his potential customers are aware of his existence, of course. Budget limitations of the small studio permit only minimal advertising. Therefore, one must show his ability by turning out a product of sufficiently high quality that word-of-mouth and advertising will do the trick. Æ

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Audio Testing in a Broadcast Studio*

FRED L. ZELLNER
American Broadcasting Company

FOR SOME TIME, there has been a certain amount of confusion with regard to level measurements in VU, DBM, PROGRAM LEVEL, PLUS 8, TEST LEVEL, etc.

A DECIBEL (dB) is a unit of measurement which represents a ratio of two (2) powers. It is usually used to indicate the difference between an input and an output, a before and after, a yesterday and today. Just as it might be said that the temperature has risen 4 degrees during the morning, so it can be said that the power in a particular circuit has increased 4 dB, or 10 dB, etc. Just as in using a thermometer, there arose a need to have some reference point, such as zero; this same need arises when using decibels. In the case of the thermometer, there are several reference points. There is one for the CENTIGRADE thermometer, which is set at the freezing point of fresh water. There is a different zero for the FAHRENHEIT scale, based on the temperature of freezing a salt solution. The KELVIN scale has still another point for its zero (probably the most logical, as well as the most useful, scientifically), based on the temperature at which all molecular motion in an object ceases.

Similarly there are several reference points when it comes to a scale based on decibels. Acoustically, a zero reference was set at the threshold of audibility. But when it came to electronic equipment using loudspeakers to overcome the background level in the average living room, this zero was found to be way up at 40 dB on this acoustical scale. Therefore, an arbitrarily set value of 6 milliwatts of power dissipated across 500 ohms was set as the reference power, and this level became known as 0 dB.

All was well and good until it came to measuring complex waveforms such as those produced by normal speech and music. No longer did RMS value have a meaning. The average power level of speech and music is much lower, compared to the peak value, than for a corresponding sine wave with the same peak. Sharp, thin peaks are the rule—greater spacing between peak pulses is produced—in short, sinusoidal measuring techniques no longer apply. Therefore, a special meter was developed (with certain specific damping characteristics) which would respond in a definite manner to these complex waveforms. This meter read a sort of average audio power (as well as reading the RMS value if a sine wave of 1 kHz was used). It required its own zero. An altogether new reference was used: namely, 1 milliwatt of power across 600 ohms, using a frequency of 1000 Hz.

Since this was a meter which measured the "volume of audio," the unit of measurement became known as a "VOLUME UNIT," or VU. If a 1000-Hz sine wave is measured by a VU meter (or VI, for Volume Indicator), the RMS value of the voltage is the result. A 1000-Hz sinewave with power of 1 milliwatt across 600 ohms will measure 0 VU.

By Ohm's Law, we find that it represents an RMS voltage of 0.775 volts (across 600 ohms):
$P = E^2/R = 0.775^2/600$ (watts) = 1/1000 (watts) = 1 mW.

About this time, it was decided, for the sake of conformity, to redetermine the standard dB, and use this same 1-mW base instead of 6 mWs. In order to distinguish between the two standards, the power ratio using 1-mW base was designated a "dBm" (the "m" standing for milliwatt). The old designation of 0 dB has practically been eliminated, and the new reference of 0 dBm is almost universally used. With a sine wave, then, VU and dBm mean the same thing. However, dBm is used ONLY with sinewave signals, and NEVER with audio.

When program audio is sent through a circuit terminated in 600 ohms, the level can be read as a certain number of VU above or below the standard 0 VU which has been established. It has become more or less standard practice for U. S. radio stations to use +8 VU as the standard for feeding telephone lines, transmitter lines, etc. If a meter attenuator is inserted in the metering circuit, and set for +8, then when the meter peaks zero, the VOLUME UNIT of that peak is in reality 8 VOLUME UNITS above the standard reference. It is called "zero at +8," and is, in reality, +8 VU. If a 1000-Hz sinewave is used, a VU is the same size as a dB, and 0 VU and 0 dBm are then the exact same level.

Actually, the restrictions (600 ohms, 1000-Hz, etc.) are for the VU and not the dBm. The dBm is a unit of finite audio power, when using sine-wave signals, which has as its zero reference a power of 1 mW. This can be across any impedance whatsoever. One mW across a 4-ohm speaker coil or 1 mW across a ½-meg grid resistor would each represent 0 dBm, although the voltages would be widely different. Thus, the meter would have to be calibrated in volts, and the value of the power in dBm calculated.

*Original paper was presented at the NAB Engineering Conference, April 1-3, 1968. A complete transcript of the technical papers presented at the conference plus a transcript of the FCC/Industry Panel discussion is available in a 256-page, plastic-bound book, Technical Papers Presented at the 1968 NAB Engineering Conference, $10.00, published by TAB Books, Blue Ridge Summit, Pa. 17214.

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The VU on the other hand is tied to a 600-ohm terminated circuit because it uses a specific meter calibrated in VU (and thus, in a sense, in power units). Actually, it is a voltmeter and will read the actual voltage across the two points being measured, but it assumes that the impedance is 600 ohms when it does the mathematics necessary and reads out in terms of power (VU). Even a meter which is calibrated in dBm, and reads dBm directly, must be used in a circuit with the impedance specified for the particular meter. This is usually 600 ohms, but does not have to be. In the case of the VU meter, the impedance must always be 600 ohms because that is one of the specifications of the meter. Thus, if a 600-ohm dBm meter is compared with a VU meter, and sine waves are used, they will both indicate the same.

Now a VU meter is so constructed and damped that a sharp audio peak cannot possibly be followed by the meter. In fact, the meter presents a sort of average of the audio, and the "peak" that is seen and read on the meter is only a fraction of the actual voltage that was present to produce the peak. This produces a very important concept.

The statement has often been made that "audio peaks are higher than sine-wave peaks for the same VU meter reading." In fact, this statement is made frequently, but without a picture in the person's mind of what the real significance is. It has also been said that there is a difference of about 7 dB between the two. (Some say 10 dB, others from 8 to 14 dB.) In order to illustrate this, an oscilloscope was set up and pictures taken of the phenomenon under discussion.

No hard fast rule can be laid down with regard to the real amplitude of audio peaks, because so much depends upon the tone being used, the vowel sounds, and the particular pitch or quality of a musical note. Again, it is a matter of averages. The photographs shown here were taken with slight time exposures, less than a second, so that there was an average of many tones and syllables in each case. First, a tone was sent through some equipment so that the VU meter read zero at +8 VU. (This was also +8 dBm.) The signal was fed into an oscilloscope, and the vertical gain adjusted to obtain 2 cms of deflection. The trace was photographed. A card, cut to size, was then placed over the oscilloscope screen so that the sinewave was completely covered. (In this way, the next trace would not obliterate the sinewave on the photograph, since a double exposure was planned.)

Audio was then picked up from the radio station, and an audio man "rode gain" as he would have for a program, aiming for a good ZERO LEVEL (at +8, of course). The resulting trace was then double-exposed on top of the previously photographed sinewave. Only that portion of the audio which was greater in amplitude than the sinewave was visible because of the card covering the center portion of the oscilloscope. The spectacular result is shown in Fig. 2A.

The 2 cms represents the reference level, and it can be seen that audio peaks cover at least a 5.6-cm range, representing a voltage 2.8 times the voltage of the reference signal. This translates into 9 dB.

Figure 2B is another picture of the same thing, but with different program material. The ratio here calculates to 10.8 dB. Occasionally during this experiment, peaks as high as 14 dB above reference 0 were noted, although not caught on a photo. This average value of 10 dB or so is what is termed HEADROOM.

Thus, as a VU meter is read showing a tone at 0 VU, and then program audio peaks hit this same 0

Fig. 1—Typical audio equipment block diagram shows test levels.

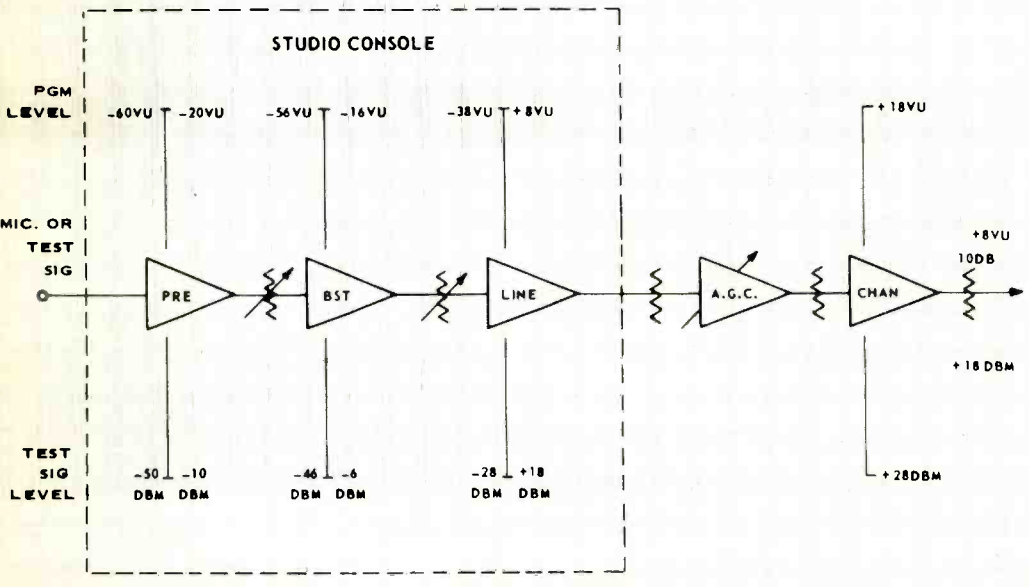
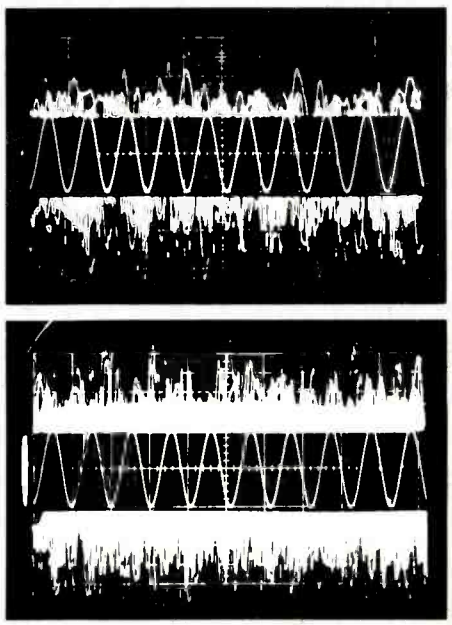


Fig. 2—Double exposures show relationship between sine wave and audio levels.



VU, it must be remembered that the peaks that cause the 0 hits are in reality approximately 10 dB higher in voltage, or approximately 3½ times the voltage of the steady-state tone that also reads 0 VU.

In checking out the studio, we are primarily interested in determining the distortion produced by the system, its overall frequency response, and how far below program level is the basic noise of the system. The first question which arises for all three tests is to determine the level at which the tests are to be made. Since the tests will be made using a steady-state tone, we realize immediately that we will not be checking out the system properly unless we do something about checking at the peak-to-peak amplitude that we have found audio peaks actually run.

Since we say that there is about a 10-dB discrepancy, it has become customary to run all studio checks at a level which is 10-dB higher than normal program levels. This means +28 dBm where a normal program level would be +18 dBm, or +18 dBm where normal program level is +8 dBm, or −50 dBm where normal program level (microphone) would be −60 dBm, etc.

The actual test lineup of amplifiers and equipment will vary from station to station, but let us use Fig. 1 as an example, showing a preamp for the mike, followed, after suitable attenuators for control, by a booster amplifier, and finally a line amplifier for the studio output. This would then go through an AGC amplifier, and a line amplifier for transmission to the telephone company or the transmitter.

Assume for purposes of discussion that average mike levels is −60 VU (this also happens to be a good standard to use). Thus, to duplicate program meter readings, a tone at the input to the preamp, or −60 dBm, would be used. (0 dBm is produced by 0.775 volts across 600 ohms, or by 0.387 volts across 150 ohms. A signal 60 dB less than this would be .000387 volts across 150 ohms.)

Set up a signal generator at 400 Hz and −60 dBm at 150 ohms output impedance, which is the usual input impedance, and leaving the master volume control where it is usually used by the audio man; set the microphone attenuator so that the output of the studio is normal program level, as shown on the VU meter. (This is usually 0 at 8 VU.) This is then metered after the final channel amplifier and either before or after the final splitting pad. (See Fig. 2.) Let us look at it BEFORE the final 10-dB pad which normally feeds the phone line, monitors, etc. At this point, then, the level will be plus 18 dBm. (After padding, the normal +8 will appear for transmission over the line.)

Now comes the big moment! Increase the signal generator output by 10 dB (to −50 dBm). This is to make the sine-wave peaks more closely resemble the actual audio peaks that are normally used through these circuits. See Fig. 3.

The needle of the VU meter in the studio console will "wham" against the pin, but have no fears or misgivings. Do NOT change the attenuator or remove the meter from the circuit. I have been doing this at regular intervals for several years now, and although I had strange feelings the first few times, I soon found that the meter is built to withstand this mistreatment. In fact, it can withstand this level continuously for hours. The output level at our measuring point is now +28 dBm. *All studio tests are made at this level.*

It is not intended in this article to describe methods of making studio or transmitter proof of performance tests, but rather to point out some interesting traps and phenomena that have been observed during several years of making these proofs at regular inervals.

Fig. 3—Photographs show sine-wave level increased to closely resemble audio peaks.

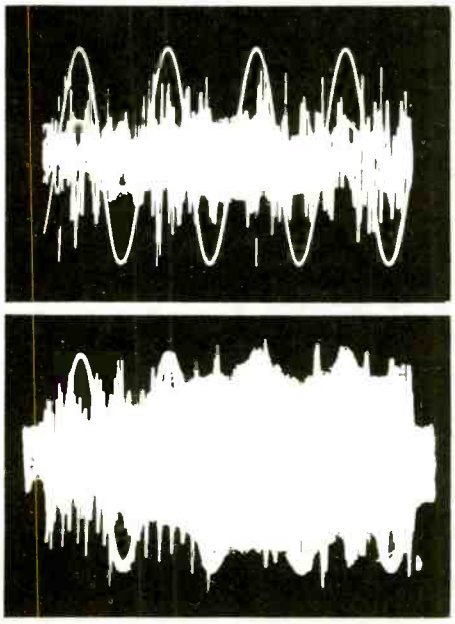


Fig. 4—Photograph (A) shows noise level at 55-dB below normal program level. (B) Illustrates distortion at same level.

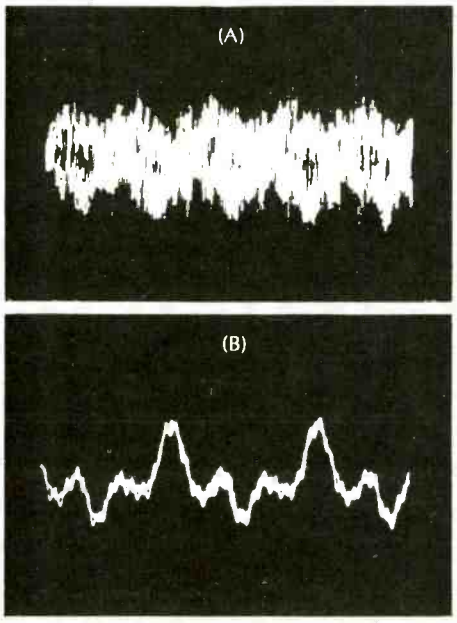
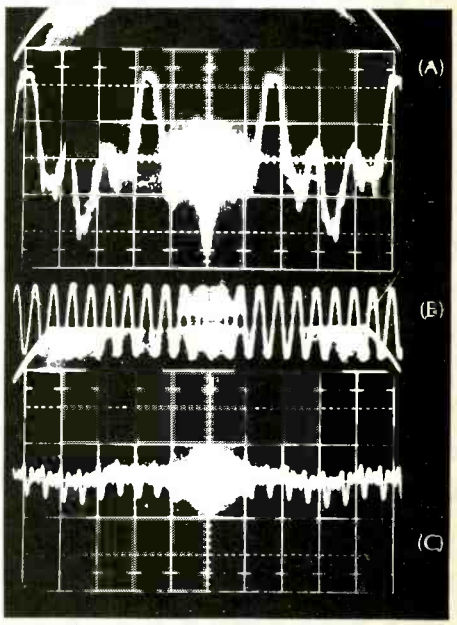


Fig. 5—(A) Distortion signal from Fig. 4B. (B) Test frequency. (C) After a 40-dB pad was inserted.



To make noise measurements, the reference level is taken at the +28 dBm point. Using this as reference, the line is terminated at the input jack field, and using a Noise and Distortion Analyzer, the noise level is determined. If everything is working properly, the preamp will be the determining factor in the noise level. To pass studio specifications, the noise should be 65 dB below the reference of +28 dBm. (This, of course, converts to −37 dBm.) It also means that the noise is 55 dB below normal program level. Fig. 4A shows the nature of the noise that is present at this low level. (In this particular case, the noise measured 72 dB below reference.) The overall patterns shows some semblance of 120 Hz. The fine peaks, of course, are a random noise caused within the preamp's first stage and/or the input transformer.

In making distortion measurements, again the +28 dBm level is used as a reference, and the distortion analyzer removes the original signal (400 Hz, for example); what is left is read as distortion. Fig. 4B shows this distortion. In this case, it measures 0.9%. The basic configuration is 180 Hz.

There is something interesting about this measurement of the distortion. Since it was difficult to understand where a 180-Hz signal could be originating in the studio audio equipment, I suspected that it might be coming from the signal generator itself. This indeed proved to be the case. The 180-Hz signal is apparently entering the system someplace in the output transformer of the generator. There is a calibrated ladder attenuator preceding the output transformer. It was found that by inserting a 40-dB pad between the generator and the audio jack field, and then increasing the signal to the output transformer by 40 dB by means of the ladder attenuator, it became possible to attenuate the 180-Hz signal, while keeping the signal strength of the signal the same, as far as the studio was concerned. Fig. 5A shows the distortion signal again, as shown in the last photograph. Fig. 5B represents the frequency that was used (400 Hz). Fig. 5C (to the same scale as Fig. 5A) shows the result after the 40-dB pad was inserted. The signal reference was the same, but the distortion now measured only 0.15% (a decrease of 15 dB). The distortion consists mainly now of third harmonic of 400 Hz, or 1200 Hz. Thus, the studio was really cleaner than the tests at first seemed to indicate —an interesting situation!

There are one or two interesting points that should be brought out when proofs are made through both the studio and the transmitter. Again, the matter of the level which is used is a matter of importance. If the level of the signal into the studio is decreased to the program level of +8 dBm and sent to the transmitter, it will be found that the modulation of the transmitter will be about 45%, or approximately 7 dB below 100%. The transmitter is a "peak-sensitive" device, and 100% modulation is determined by the actual peaks. Thus, a similar level sine wave (as measured on a meter) would be about 7 dB lower on peaks. Obviously, the test level must be brought up to 100% modulation. The question arises as to *where* the level should be brought up to produce 100% modulation. The limiter at the transmitter has been disabled. Let us see what would happen if the level were to be brought up by means of the amplifier in the limiter.

Let us assume, for example, that our studio barely made its noise specification of 65 dB below reference +28 dBm. Since we are now using a signal of 10 dB lower than this, or +18 dBm at our measuring point (+8 dBm at the input to the telephone line to the transmitter), the noise would now appear to be only 55 dB below this new reference. If the gain of the amplifier at the transmitter is now raised to produce the 100% modulation, both signal *and* noise from the studio are raised, and the relationship, or 55 dB, remains the same.

Let us suppose further that the transmitter itself (FM) also just barely makes its specification of 60 dB below reference 100% modulation. The resulting overall noise through the entire system would then be about 53½ to 54 dB below 100% modulation, and this would be far from the FCC specification of 60.

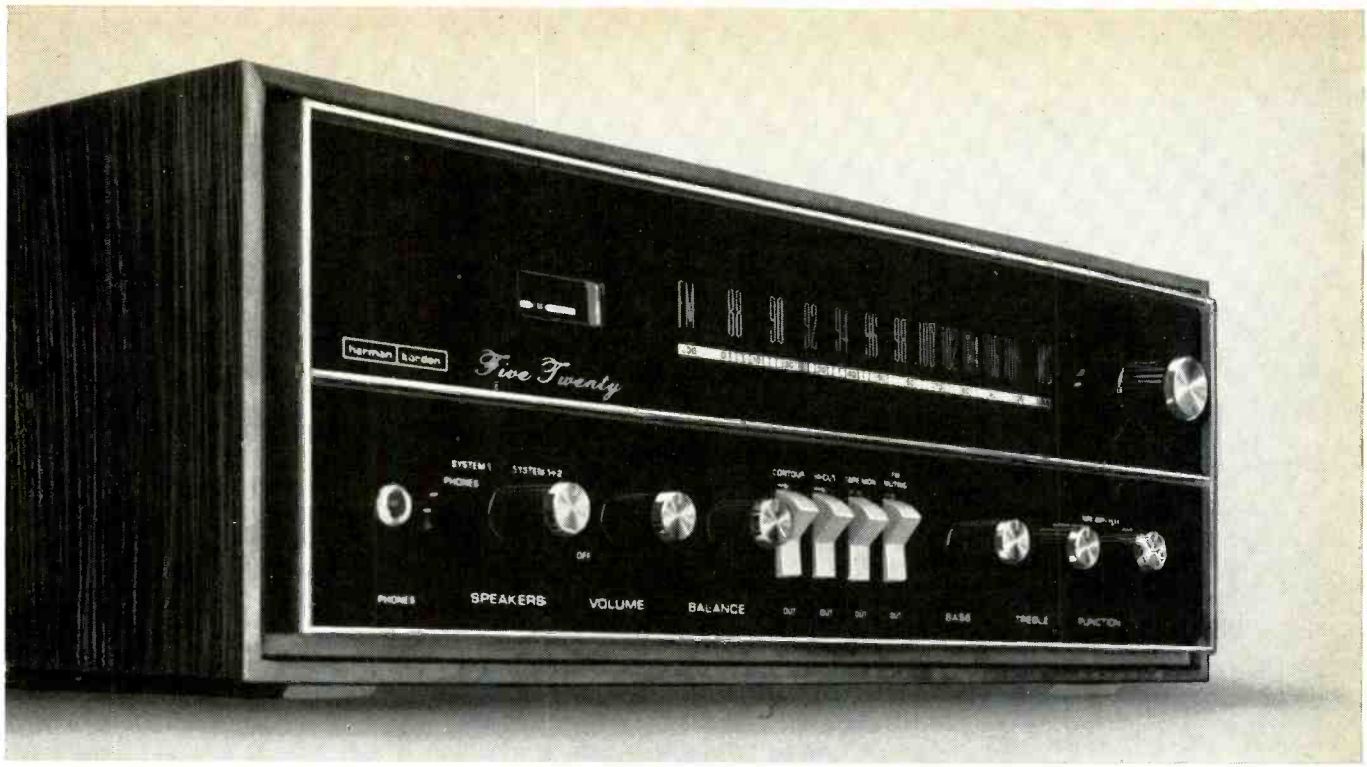
Even if two identical signals of −60 were added, the sum would be only −56 dB. (I know the books say it should be −57, but I have never found this to be the case in actual practice.) Similarly, −63 plus −60 equals about −58½, etc. Finally, when one noise signal is about 6 dB or farther from the other one, there is no apparent addition; (i.e., 66 plus 60, or 68 plus 62, etc.).

This then gives a clue as to where the signal should be raised to produce the 100% modulation. With the generator in the studio set for −60 dBm, and no controls touched at the transmitter, except for disabling the limiting action of the limiter, the modulation of the transmitter should be brought up to 100% by increasing the signal going into the preamp. It will be found that this increase is usually in the nature of 7 dB (achieved by adjustment of the ladder attenuator on the signal generator) so that the studio signal-to-noise ratio in the extreme case mentioned above would become about 62 dB below the reference studio output. This −62 added to −60, although still not passing specification, would be about −58 dB.

This shows that a better than minimum noise measurement must be met in the studio and transmitter in order to pass the overall specifications.

This level of signal then is used throughout the overall tests through studio and transmitter. The 40-dB pad is again used for the distortion measurements only. During the frequency-response test, the signal is increased or decreased at the input to the studio for each frequency to allow for the pre-emphasis at the transmitter for various frequencies.

Thus, we see that certain precautions must be taken when making tests of audio equipment. HEADROOM must be allowed for and taken into consideration. The terminology VU, DBM, etc., must be carefully understood so that there is unformity of thought and expression during tests. All in all, it is a very intriguing subject, and one in which some careful study and consideration will pay dividends in better audio and smoother operation, and greater satisfaction for the operating personnel. Æ



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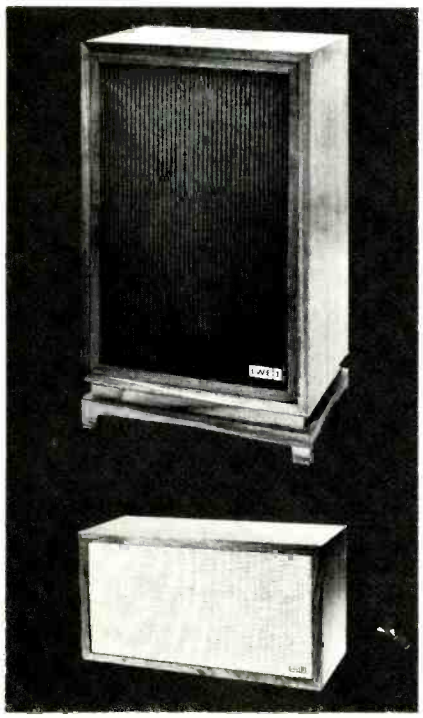
- Viking Model 433 Stereo Tape Deck
- H. H. Scott Model LT-112B-1 FM Stereo Tuner Kit

Erath Models LWE-I and LWE-VI Speaker Systems

MANUFACTURER'S SPECIFICATIONS— LWE-I: One 15" woofer, one 6" mid-range cone, one 5" horn tweeter. Nominal Impedance: 4 ohms. Freq. Resp.: (±5 dB) 22-20,000 Hz. Controls: High frequency, mid-frequency, room gain. Dimensions: 25" H, 17" W, 12" D. Weight: 61 lbs. Price: $250.00, oiled walnut; $225.00, birch; $175.00, "instant kit." LWE-VI: One 8" woofer, one 3½" tweeter. Nominal Impedance: 8 ohms. Freq. Resp.: (±5 dB) 29-15,000 Hz. Controls: High Frequency. Dimensions: 19" W, 10" H, 9" D. Weight: 46 lbs. Price: $100.00, oiled walnut; $90.00, birch; $75.00, "instant kit."

Fig. 1 – The LWE-I three-speaker system (top) and the smaller LWE-VI two-speaker system, both of which utilize a voltage feedback system.



The application of inverse feedback to amplifiers has brought them to near perfection in terms of frequency response and distortion. Therefore it is not unnatural to consider the use of feedback to improve a speaker system's performance. In fact, we heard some excellent prototypes that used the feedback principle many years ago. But none ever succeeded in production. Erath has!

Whereas others attempted to pick up the radiated sound from the speaker by means of an additional transducer—a microphone or an additional winding on the speaker voice coil—Erath speaker systems utilize an electronic network to generate their own error voltages when the speaker does not follow the amplifier output, thus compensating for any speaker deficiencies. Primarily, its purpose is to extend, electronically, the speaker's bass response, as well as to improve transient response. And whereas earlier efforts in this direction required that one purchase a complete system, from amplifiers to cabinet, the Erath speaker systems, which externally, look quite conventional, can be used with any high-quality component amplifier or receiver (the amplifier must have at least 20 dB of feedback, which most good units do).

Two of the company's four models are examined here: Model VI, the smallest in the line (19-in. x 10-in. x 9-in.) features an 8-in. woofer and a 3½-in. mid-range / high-frequency speaker: Model I (25-in. x 17-in. x 12-in.) is once removed from the biggest Erath system. It incorporates a 15-in. woofer, 6-in. mid-range, and a 5-in. horn tweeter. At $100.00 ($75.00 for an "Instant Kit") and $250.00 ($175.00 for an "Instant Kit"), respectively, they would appear to hold promise of being the most popular in the line.

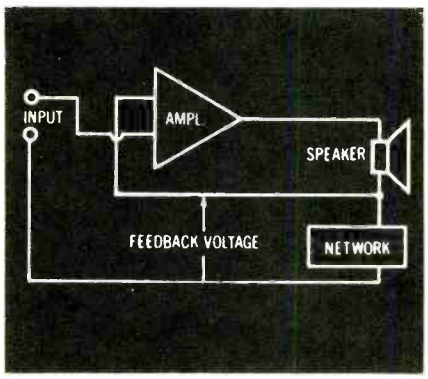
All the systems have air-tight enclosures, similar to acoustic suspension systems, but efficiency is higher than many sealed enclosure systems. Clean 20 watts/channel amplifiers should be satisfactory for normal use. The VI has a power-handling ability for program material of 25 watts (peak power may be much higher, of course), while the bigger Model I can handle 50 watts.

Whereas the Model VI has a high-frequency control, the Model I (and the other two models in the Erath line) adds a "Room Gain" control, as well as a mid-frequency control. The unusual "Room Gain" control is actually an electronic broadband filter that works in conjunction with the speaker's feedback circuit to minimize frequencies at which a room resonates. Thus, its purpose is to influence frequency and amplitude in order to compensate for room acoustic deficiencies.

Before we discuss the performance capabilities of this remarkably different speaker system, however, let us examine the theory behind it and the speaker construction more closely.

A cross-section view of any Erath speaker system would reveal a typical closed-box-enclosure construction, with lots of absorbent material—plus a "black box" and crossover networks. It's the black box that's intriguing, of course. This is the novel feedback network, which includes some reactive elements. It develops a voltage which is fed back to the same point in the user's amplifier at which the usual negative voltage feedback is introduced, resulting in the arrangement of Fig. 2. This circuit can thus eliminate boominess

Fig. 2 – The drawing below shows how feedback voltage is derived when an Erath loudspeaker/feedback network is connected to an amplifier.



of a woofer, and with proper choice of the reactive elements in the network, extend the bass response of a system appreciably.

The speaker has a 6-terminal male plug by which connections to the amplifier are made. Two of the leads carry the voice-coil current to the speaker, and two carry the "feedback" signal to the amplifier from the speaker. The other two may be used as tiepoints for additional resistors which may be needed to adapt the network to the particular amplifier or receiver being used. At the amplifier, the female output plug is connected to the speaker terminals in the usual fashion, and two additional connections are made inter-

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The popular, "ball-type" Shure Unisphere A series of microphones are the kind of public-address system problem-solvers that have made them run-away favorites with sound men, speakers, and performers from coast-to-coast. And, they are budget-priced in the bargain! They have the superb, world-renowned uniform cardioid pickup pattern that has made Shure Unidynes the industry standard for controlling feedback. Their tone is uniquely suited to natural reproduction of the entire voice range. They are unexcelled for controlling reverberant "boom" in partially filled halls. In addition, ball-shaped filter assemblies control windnoise in outdoor applications.

The new models 585SAV and 585SBV are among the most ingenious and versatile of microphones in that they feature a built-in volume control — they are ideal where amplifier controls are inaccessible, or where the speaker or singer wants to control his own volume for dramatic effects. Instantly interchangeable from stand to hand. Whatever the problem, the Unisphere A microphones can help solve it . . . economically!

UNISPHERE A

Model 585SA High Impedance; 585SB Low Impedance. Budget priced for use in low-budget systems. List Prices:
Model 585SA $65.00; Model 585SB $58.00

UNISPHERE A WITH VOLUME CONTROL

Model 585SAV High Impedance; 585SBV Low Impedance. Volume control versions of the Unisphere A. List prices for either model: $72.50

SHURE BROTHERS, INC. / 222 Hartrey Ave., Evanston, Illinois 60204

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Fig. 3

Fig. 3 – A side cutout on the LWE I shows the "black box" feedback network in the right corner.

Fig. 4

Fig. 4 – The LWE-VI is shown atop the LWE-I, both with front grille cloths removed. The LWE-I pictured here is the company's "Instant Kit," which has its controls (room gain, mid-range, high frequency) on the front panel in the event a user wishes to build the unit into the wall.

nally by means of cleverly designed "Q-ball" connectors.

This may sound like a difficult operation for the average purchaser, but Erath has made it quite simple. A complete line of adapters has been worked out by Erath engineers, and for $5.00 you obtain the one to match your amplifier or receiver. These simply clip onto the designated points in the amplifier at the feedback junction, in accordance with the instructions accompanying the adapter. Illustrations of the proper connection points for most popular amplifiers and receivers make this quite simple to do.

The units we tested were used with a Sony TA-1120 amplifier, and we also obtained the adapter for the Acoustic Research amplifier. The Sony had two resistors and a choke, while the AR had no external components whatever —simply the cable plug with four leads coming out from it—two for the speaker terminals on the amplifier, and two to be clipped on resistors inside the amplifier.

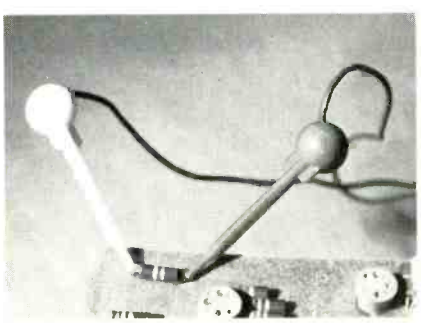
Performance

Listening to the small, though not subminiature Erath VI reproducing a variety of source material, we were impressed by two facts: (1) For a small, two-way system, it exhibited smooth overall response, and (2) bass response was exceptionally good.

Concerning the latter, the immediate effect was very much like hearing an unexpected half-octave more of clean bass, not to mention the outstanding crispness of the low end in general. Our optimum preamp control settings for this speaker (s) was bass tone controls slightly on the minus (–) side, treble controls normal. There was a whisper of "tightness" while reproducing low frequencies, but no tubbiness at all. Mid-range and treble response were non-peaky and well balanced, with sufficient high-frequency dispersion for small-to-medium-size rooms. Judging the merits of the Model VI as a small bookshelf speaker system, we consider it to be among the best we've heard to date. It appears to meet the manufacturer's specification claims, and its low-frequency performance rivals that of many larger and more expensive speaker systems.

Erath's Model I, a large bookshelf-size or medium floor-standing speaker system (a $12.00 walnut base is available for the latter purpose), sounds better and better as one listens to it (them) for a substantial period of time. This is characteristic of truly fine speaker systems. With nothing flashy to deceive a person on an initial hearing—certain frequencies favored, for example—one cannot quickly focus on an outstanding attribute. The Erath I is such a speaker system. It does not noticeably favor any part of the frequency spectrum, and the frequency range is extremely wide and marvelously balanced. The upshot of this is a speaker system whose acoustic reproduction does not appear to be entirely within the environs of its enclosure. Thus it produces a wide, open sound: a transparency that is most appealing.

Fig. 5 – Erath's cleverly designed adapter clips onto the feedback junction point in an amplifier.



The Model I has a control labeled "room gain" which compensates for different sizes of listening rooms. One of our listening rooms is not large, being about 3200 cu. ft., but with an additional 1400 cu. ft. opening off it—the kitchen and a short hallway. The main portion of the room is an approximately symmetrical "L" shape. It should develop resonances at about 26, 44, 61, and 77 Hz, thus coloring any sound reproduction to some extent. The effect of the "room gain" control is to use these resonances—since they influence the radiation resistance of the speakers—to control the amplifier and to make it perform in a way which will minimize the resonances. This is, to our ears, just what happens. Low-frequency resonances due to the room dimensions cannot be damped out by the usual acoustical materials but the LWE speaker can provide a controlling signal for the amplifier so that the final result is a great reduction in resonance effects.

By measurement, the response in the air rose slightly at 40 Hz, then began to fall off in accordance with the roll-off network in the Sony amplifier—a bridge-T arrangement which starts to cut off about 60 Hz, and is down 6 dB at 10 Hz.

There is no doubt that a three-way, three-speaker system, which includes a 15-in. woofer, has the capability of producing fine sound. But since the Model I goes beyond what is expected of it, we can only attribute this to the feedback network developed and applied by the L. W. Erath Company. Without equivocation, the Model I exhibits immaculate low and very-low bass response. We've never heard better in this size enclosure. Mid-range and high frequencies are also excellent, with the latter having the characteristic brilliance of a horn tweeter.

Clearly, speaker system designs are not nearly as traditional as one might think they are. Proof of this is the new design path taken by Erath in developing a frequency-selective feedback system that automatically makes a high-quality amplifier feed a corrective voltage to the speaker when it fails to follow the amplifier's output faithfully. More importantly, both LWE feedback speaker systems examined here proved to be excellent performers —especially in the areas of frequency response and transient response.

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The Sony Side of the Street

(it's any place they're showing the new 6060 receiver).

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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MANUFACTURER'S SPECIFICATIONS— FM Section: IHF Sensitivity: 1.7 μV. Image Rejection: 76 dB (@ 98 MHz). Signal-to-Noise: 65 dB. Capture Ratio: 1 dB. Stereo Separation: 37 dB (@ 1 kHz). AM Section: IHF Sensitivity: 20 μV. Image Rejection: 60 dB (@ 1 MHz). Amplifier Section: Music Power (total): 170 watts @ 4 ohms. 140 watts @ 8 ohms. RMS Power (per channel): 60 watts @ 4 ohms. 55 watts @ 8 ohms. Harmonic Distortion: Less than 0.5% at rated output (1 kHz). Frequency Response: 20 Hz to 70 kHz ±1 dB. Power Bandwidth: 20 Hz to 70 kHz. Hum and Noise: Tape Head: better than 75 dB. Mag. In.: better than 80 dB. Aux. In: better than 90 dB. Damping Factor: 25 @ 8 ohms, 1 kHz. Tone Controls: Bass +12 dB, −14 dB @ 50 Hz; Treble +10 dB, −11.5 dB @ 10 kHz. Filter Action: Low: −6 dB @ 50 Hz. High: −10 dB @ 10 kHz. General: Dimensions: 16 7/16 in. W x 5 1/2 in. H x 13 3/4 in. D. Price: $360.00 (Including Metal Cover and Walnut-Finish side panels).

Among the better, high-powered crop of receivers offered to the stereo component purchaser this year is the Pioneer Model SX-1500T. It is equipped with just about every control one might expect to find on separate amplifiers or preamplifiers.

The unit is well-styled, sporting a light gold and black anodized heavy front panel, a walnut-colored metal cover and two walnut-finished side panels, all of which combine to give the unit a trim, modern look. The upper portion of the panel contains the tuning dial area, including the usual stereo indicator light and a peak-reading tuning meter. A 0-100 logging scale serves to divide the upper FM scale from the AM scale beneath it. Calibration markings are not pinpointed, however (while numerals appear for every 2 MHz of FM dial spread, no definitive marking point for each numeral is present). One can use the logging scale, of course.

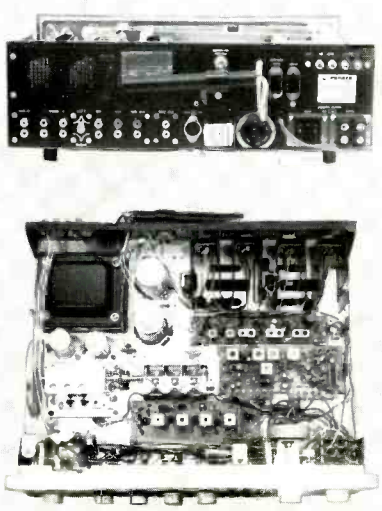
To the right of the tuning dial area are the flywheel-mounted tuning knob (not quite velvet smooth, but satisfactory nonetheless) and the main function selector switch. Besides the expected positions of this switch, such as AM, FM MONO, FM AUTOMATIC STEREO, PHONO, and AUX, there are two very welcome ones — tape-head positions equalized for either 7½ ips or 3¾ ips for direct playback from a tape transport. Few receivers, let alone amplifiers or preamplifiers, provide this option and, we might add, both of these positions *do* offer correctly equalized (NAB) response within 1 dB from 30 Hz to 18,000 Hz.

The lower portion of the panel, starting at the left, contains a speaker selector switch (main, extra, both, and speakers "off") which, in its counter-clockwise position, turns off all power to the unit. Bass and treble controls are of the dual concentric type, with clutch action for left and right channels.

These are followed by conventional balance and volume controls. Secondary functions are controlled by means of six small, elegant lever switches. These functions include loudness contour, low filter, high filter, muting, AFC, and a "Phono-Phono 2" switch to enable the user to have, say, a turntable and a record changer connected to the equipment simultaneously. Again, this last is a feature not normally found on all-in-one receivers. The usual phono jack and a rotary six-position "mode & tape monitor" switch completes the layout of the front panel. Why six positions for "tape monitoring"? To enable you to either *listen* to left, right, or stereo channels through both speakers or to *monitor* left, right, or both channels from a "three-headed" tape recorder. Figure 1 illustrates the front of this receiver.

Fig. 2 (upper)—Rear panel layout of the Receiver.

Fig. 3 (lower) — The top-of-the-chassis view shows the clean construction of the Pioneer unit.



Besides the usual input and recorder output jacks, the rear panel is equipped with two a.c. convenience outlets, the necessary antenna and ground terminals, the fuse post (as well as the voltage selector in the case of the SX-1500TF model, which features selectable voltages from 110 V to 240 V) and a loop-stick AM antenna, movable for best AM reception. Speaker connection is unique. There are four *polarized* two-prong receptacles, one for main speaker connection and one for extra speaker connection. Four two-prong plugs are supplied separately, to which speaker leads can be connected under the heads of screws in the usual fashion. These plugs are then inserted in the appropriate sockets. This has the advantage that once you have properly wired the speakers to the plugs for correct phasing, speaker systems may be disconnected at any future time (for moving, cleaning, etc.) and, when reconnected, phasing will always be correct, thanks to the polarized plug and jack combinations.

The rear panel layout is shown in Fig. 2, while Fig. 3 is a close-up view of the chassis surface, showing the separate AM and FM front ends, as well as three of the *eight* P.C. boards which go to make up this unit. The four power output transistors are mounted on large, heavy heat sinks to which are affixed heat-sensitive thermistors for bias stabilization of the output circuits. Electronic switching in the power supply further protects the output transistors from inadvertent shorting of speaker leads and other excessive current loads. In all, this husky receiver contains 4 integrated circuits (IC's), all in the FM i.f. strip, 1 FET (in the FM front end or r.f. section), 39 transistors and 29 diodes.

Performance

Measured performance was generally excellent, adhering quite closely to published specifications. Referring to

All turntables are not created equal.

(This is a public service message from Marantz.)

There are two ways to build a turntable. The ordinary way. And the Marantz straight-line tracking way.

Straight-line tracking makes a home turntable system reproduce the sound on a phonograph record exactly as it was originally etched by the studio cutting head. And only Marantz has straight-line tracking. Straight-line tracking keeps the tone-arm precisely tangent to the grooves—not sloshing around in them.

That's why it is the *only known way* to give you absolutely uniform stereo separation and frequency response from the outermost groove to the innermost (where distortion is greatest). In addition, straight-line tracking eliminates tracking error distortion, uneven stylus wear, and skating force.

Another Marantz feature, positive cueing control, ends accidental record scratching forever. One simple control knob lets you set the stylus in any groove you desire.

The Marantz Model SLT-12U turntable is equipped with a universal pick-up head which is adaptable to a broad selection of popular cartridges. No wonder—feature for feature—it is the ideal instrument to enable you to enjoy perfect stereo sound in your home—exactly as heard in the finest recording studios. And best of all, it is priced at just $295.

There is so much that goes into making a Marantz a Marantz, that your local franchised Marantz dealer will be pleased to give you a demonstration. Then let your ears make up your mind.

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Designed to be number one in performance... not sales.

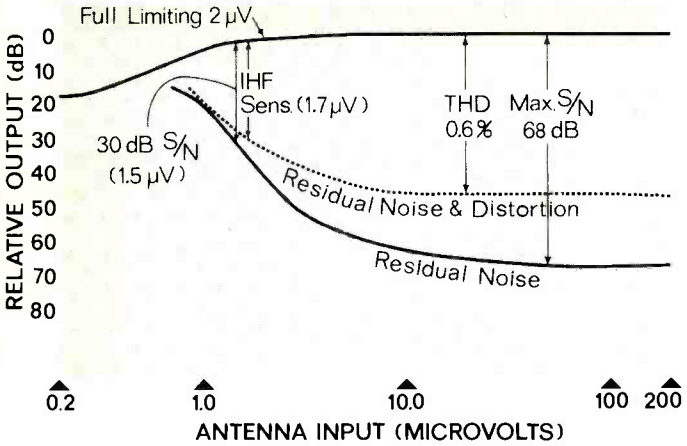

Fig. 4—FM performance characteristics of the SX-1500T.

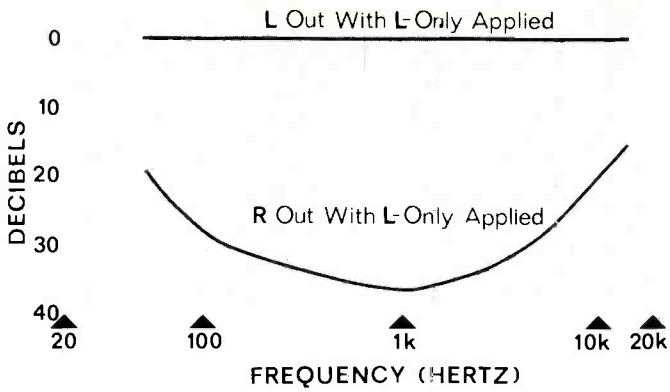

Fig. 5—Stereo separation characteristics of the Pioneer receiver.

Fig. 6—IM and THD characteristics. Note minimal distortion at low power output, indicating virtual absence of crossover distortion.

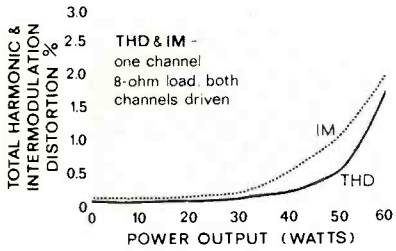

Fig. 7—Power-bandwidth curve shows excellent performance.

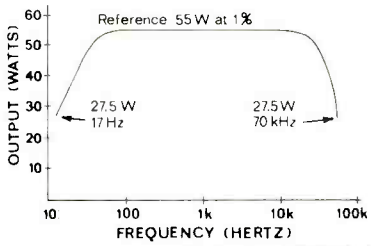

Fig. 8—Tone-control and HI and LO filter characteristics.

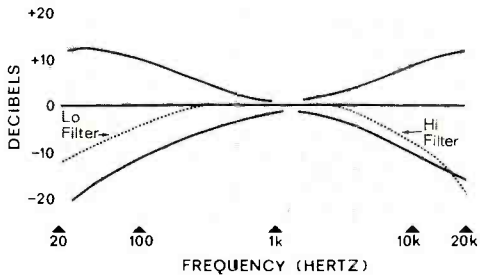

Fig. 4, you can see that FM IHF sensitivity was exactly 1.7 μV, as specified. Ultimate S/N ratio was 68 dB in FM, as compared with a claim of 65 dB. Total harmonic distortion at 75-kHz deviation was a mere 0.6%. In stereo FM, for equivalent modulation, distortion measured 0.9%, after eliminating residual 38 kHz and its harmonics. These residual signals were down some 36 dB at the tape output, a figure that could stand some improvement. The use of IC's in the IF strip provides excellent limiting (within 1 dB of full output at only 2 μV), while the FET in the r.f. section accounts for the very excellent image rejection observed as well as the almost complete absence of measurable cross-modulation. As for stereo FM performance, switching from mono to stereo was accomplished with no accompanying clicks or noise bursts. Stereo separation, as shown in Fig. 5, was 37 dB at 1 kHz and no worse than 20 dB at any frequency from 50 Hz to 10 kHz.

As for audio performance, we measured 52 watts rms per channel at a THD of 0.5%, only a fraction short of the 55 watts spec. In any event, we were able to pump 55 watts out per channel at a total distortion of only 1%, as shown in Fig. 6. IM distortion was 1% for 50 watts rms per channel output, and only 2% at a whopping 60 watts out.

The engineers at Pioneer must belong to the "wide-band" response school for, although we suspected that the Power Bandwidth published specification might be a misprint, it actually *does* extend from 17 Hz (they claim only 20 Hz) to 70 kHz! You'll never lack for "highs" with this one! Power Bandwidth is plotted in Fig. 7.

Tone control range is as shown in Fig. 8. Also in the same figure are the "cut" characteristics of the low- and high-filters. These filters have pretty much the same slope as the tone controls (about 6 dB/octave) even though the crossover points are a bit lower and higher, respectively. Had they been designed with a 12 dB/octave slope, they might have proved more useful. With tone controls set for flat response and no filters in the circuit, one can sense the smoothness of response and excellent transient attack characteristics of this receiver. We borrowed an extra pair of low-efficiency, high-quality bookshelf speakers (in addition to the main pair) to see what 55 watts can do, and we can report that this amount of power was more than adequate for driving *four* low-efficiency speakers. The sound remained clean from the lowest level to "3 o'clock" on the volume control.

With a sensitivity of 1.7 μV on FM, it goes without saying that we pulled in just about every mono and stereo station from "fringe" and locally. The muting was effective between stations and added no distortion on marginally received signals. Level balance between AM, FM, and Phono was very good, but Tape Head inputs had a bit more gain, so volume had to be lowered when switching to this service. There is a separation control on the rear panel which has been factory preset and customers are warned not to change the setting (so why make it accessible?), but aside from this paradox, controls handle well, are smooth to the touch and very functionally arranged.

If you crave lots of power and don't want to get involved with separate preamp-amps and tuners, the Pioneer SX-1500T AM/FM stereo receiver certainly has enough power and enough true component features to make it very worthy of consideration at its remarkably low price of $360.00.

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This beautiful four-headed monster does away with amateurs.

Once you've met up with our monster with four heads, you're done for. Your amateur days are over. That's because the 4-track Solid-State stereo RS-790S has just about everything you need to do a professional job of taping.

First, there's 3-speed Dual Capstan drive. It ends audible flutter and wow. And the sound is all the better for it.

Four heads are better for sound, too. And the Console-Aire delivers 30-18,000 cps and a signal-to-noise ratio of more than 52 db's. It all adds up to the greater fidelity the pros listen for.

Another great thing is continuous Automatic Reverse. Records and plays back in both directions. It means no more interruptions. And you'll never have to flip over a reel again. At any point on the tape you can manually punch up reverse, too. Of course, if you don't want it to run on forever, use the automatic shutoff.

Pause Control is another nice feature. It operates in forward and reverse, and locks down for easy editing.

It gets better.

There's headphone output for private listening. Makes it easier to record sound-*on*-sound and sound-*with*-sound.

If that sounds like a lot of sound, it should. You get 20-watt output through two 7" oval speakers with baffle boards.

There's more to come. Like two Dynamic Pencil Mikes with stands. Connecting cords and other extras.

That's not all. You get 2 precision VU meters, separate tone and volume controls, lighted directional indicators, and a 4-place digital counter. Top this with a smoked-glass dust cover, and you're on your way.

After all, it's what you'd expect from the world's leading manufacturer of tape recorders.

So go into any dealer's we permit to carry Panasonic. We have a feeling that once you come face-to-face with our beautiful four-headed monster, you'll lose your amateur standing forever. (And for just $329.95.*)

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For your nearest Panasonic dealer, call (800)243-0355. In Conn., 853-3600. We pay for the call.

*Suggested list price. Canadian price higher.

Viking Model 433 4-Track Stereo Tape Deck

MANUFACTURER'S SPECIFICATIONS— Record/Playback Frequency Response: 40 to 18,000 Hz ±3 dB @ 7½ ips, 40 to 12,000 Hz ±3dB @ 3¾ ips, 50 to 6,000 Hz @ 1⅞ ips. Signal-to-Noise Ratio: 54 dB or better. Distortion: 1% THD at 1000 Hz. Bias/Erase Oscillator: 80 kHz. Crosstalk Rejection: 55 dB. Fast Forward/Rewind Time: 70 sec. for 1200 ft. Reel Size: 7" max. Wow and Flutter: Less than 0.2% @ 7½ ips, 0.25% at 3¾ ips, 0.5% at 1⅞ ips. Diemnsions: 15¾" W x 14⅜" H x 8¾" D (behind panel, 6½"). Price: $369.95. With walnut base, $389.95.

Viking's new Model 433 is an attractively styled, unusually versatile, medium-priced machine. It offers lots of worthwhile features for the serious tape recordist — sound-on-sound and echo, among them. At the same time, it is an impressive performer. The model number "433" spells out what the recorder includes: four tracks, three hyperbolic tape heads, and three speeds. (It also has three motors.) As distinguished from a tape recorder system, the Viking deck does not incorporate power amplifiers and/or speakers, of course. Thus, it is not designed to be used as a portable unit (though it could be used for this purpose with the addition of portable amplifiers and speakers such as Viking's Model 4400 speaker/amplifier system).

The 433 deck, which is a gold and black unit, can be purchased for cabinet or panel-cutout "custom" installation, or with an oiled-walnut base. The former includes mounting brackets.

An operator of the 433 tape deck is likely to find the two level meters most useful since they are illuminated when in use, indicating the mode of operation at a glance. That is, while recording mono, only the proper level meter is active, as an example. Equally helpful are Viking's mode indicators. These are four indicators (two on each side of a level meter) that light up "Play" or "Record," with a green or red illumination, respectively. A single multifunction selector switch is used to choose the desired function which can be observed on the mode indicators (and the function read even at a 10-ft. distance).

There are separate playback level controls and monitor controls which vary their respective outputs (RCA phono jacks at the rear). A monitoring headphone jack (for 4 to 600-ohm headphones) is also provided on the front panel. The inputs have separate microphone and high-level controls so that mixing of auxiliary sources wtih microphones can be accomplished by the recorder electronics without an external mixer. The mikes plug into the front via phone plugs, while the hi-level inputs come into the rear. Another feature is an echo switch which enables some of the sound picked up by the playback head to enter the recording circuits. The record head precedes the playback head in time by a few milliseconds, so we get some echo while recording. On most recorders this requires cable patching at the rear plus some sort of control to vary the amount of echo. Here the amount of echo is controlled by the "play" controls while recording. The mode switch also includes transfer modes from left to right and vice versa, for sound-on-sound (special effects) recording. Naturally, a source/tape compare switch is also provided on the front panel.

The 433 is mechanically and electrically separable into four parts that unscrew and unplug for easy replacement in servicing. The top section has the supply and take-up reel motors, together with associated electronics. The center section contains the capstan motor and flywheel, tape-gate solenoid, and record equalization switch. A side mination, respectively. A single multisection holds the tape-motion controls and relays, and the bottom section contains all record/playback electronics. See Fig. 2.

Controls on the right consist of two mechanically interlocked, three-position lever switches which defy user error. Pause and record buttons are above the levers, and a three-position lever on the left side of the deck selects speed by moving a rubber belt between the capstan motor's stepped pulley and the flywheel. A four-digit resettable counter, above the speed selector, is driven off the supply reel. A remote pause button at the end of a long portable cord (which plugs into the rear of the deck) allows momentary pauses to be made during recording or playback.

The Viking 433 has three four-pole motors. One drives the two-pound, dynamically balanced capstan/flywheel assembly, while the others drive the supply and take-up turntables directly. A tilt-away pressure roller simplifies tape threading through the straight slot as shown in Fig. 3.

The entire head assembly (see Fig. 3) is removable by unscrewing two screws and unplugging three molded head connectors. A spring-loaded pressure pad is used at the left-hand guide and two pin guides between the erase and record heads and after the playback head, both attached to the movable gate, provide the necessary tracking for correct contact between the tape and heads. Braking is dynamic, with no mechanical brakes whatsoever. One result of this is that, in threading the tape, the reels are completely freewheeling. For someone used to mchanical brakes, there might initially be a tendency to spill tape during threading, but once familiar with reels that are completely free to turn in either direction, one can thread very quickly, with little tension applied to the tape. An idler/run-out arm acts as the tape-end and tape-break sensor, shutting everything off at the appropriate time, in addition to doing its duty as final tape guide before the take-up reel.

Seven plug-in printed circuit cards comprise the 433's all solid-state electronics. They are divided according to function, and each is removable. Playback equalization and record bias are adjustable from the rear of the deck. See Fig. 4. But since the rear cover has an interlocking a.c. power connector, a "cheater cord" is required to make these adjustments while the machine is running.

Performance

As expected, the unit performed exceedingly well throughout all our tests. Wow and flutter, measured at beginning, middle, and end of tape reel, and

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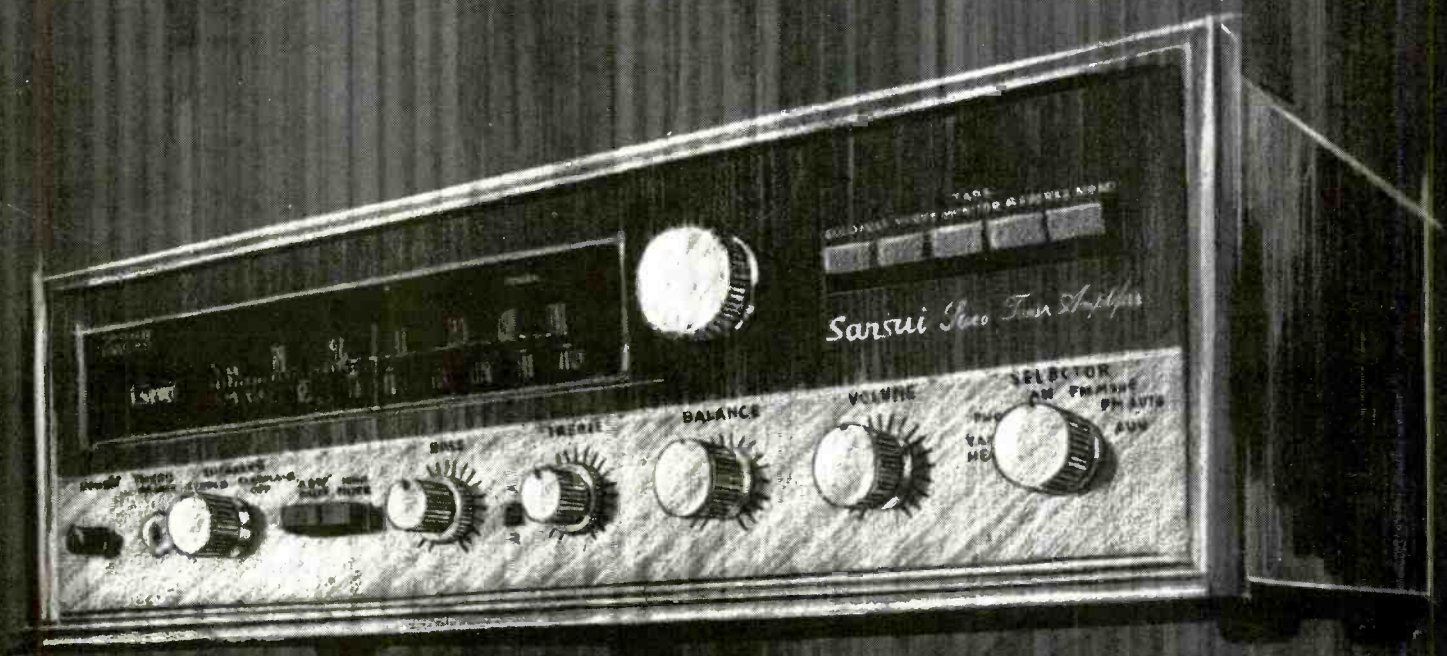

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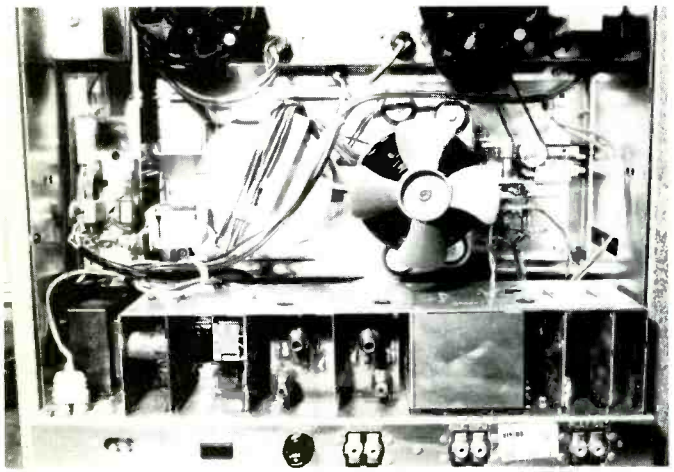


Fig. 2–Rear view of tape deck showing four demountable sections.

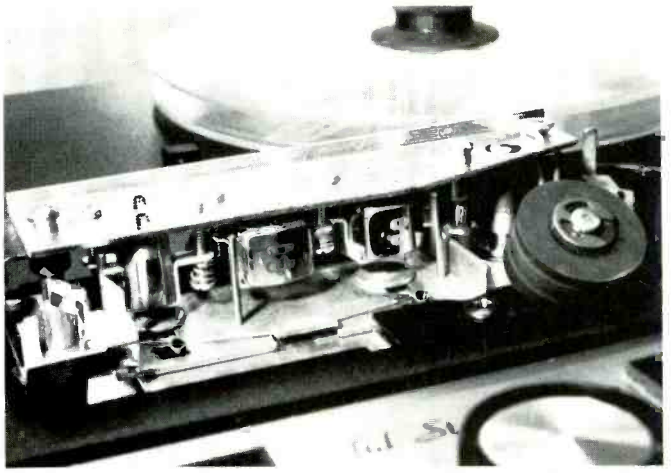


Fig. 3–Head assembly, showing capstan and tilt-away pinch roller. Note the easy access for cleaning and demagnetization.

Fig. 4–Plug-in electronics shown with metal shield removed.

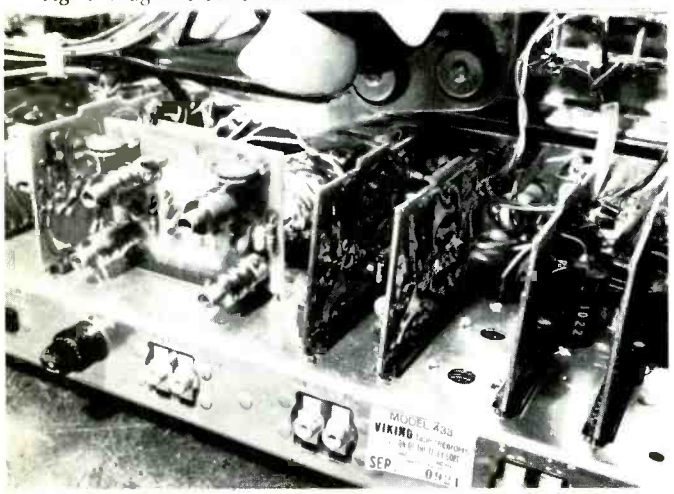


Fig. 5–Playback frequency response. Test tape was full-track, which accounts for the boost at 100 Hz at 7½ ips.

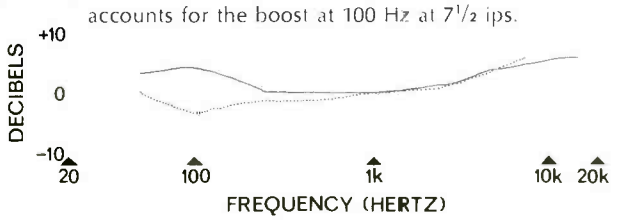
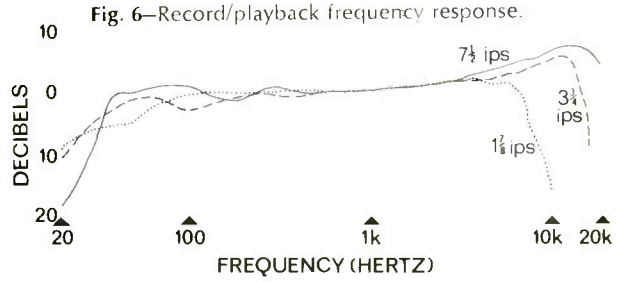


Fig. 6–Record/playback frequency response.



then averaged, was 0.14% total with 0.05% wow at 7½ ips. At 3¾ ips, the machine was better than 0.20% total with 0.12% wow. At 1⅞ ips, wow and flutter was better than 0.4 total. These figures are excellent, comfortably exceeding the manufacturer's specifications. Fast forward/rewind time was 70 seconds for 1200 feet on tape on a 7-in. reel, as specified.

The playback frequency response, as shown in Fig. 5, yields a response of 50 to 15,000 Hz ±3 dB at 7½ ips; 50 to 7,500 Hz ±4 dB at 3¾ ips. This response is very good, although the rising high end could probably be readjusted with the playback equalization control at the rear of the machine to give us a flatter curve. With the above-mentioned adjustment made, the unit is certainly suitable for high-quality reproduction of recorded tapes, being limited only by the fidelity of the source material.

The record/playback response, shown in Fig. 6, yields 40 to 20,000 Hz ±3.5 dB at 7½ ips, 40 to 15,000 Hz ±4 dB at 3¾ ips, and 50 to 8,000 Hz ±3 dB at 1⅞ ips. The response here also rises at the high end, probably due to a slight error in playback equalization. Also, Scotch 202 tape was used for our tests and, although not specified, most likely the machine was factory adjusted for the standard Scotch 111 tape, which would account for the rest of the peak at the high end. Even with the response as it stands, however, the unit meets its specifications. The record/playback signal-to-noise ratio measured 57 dB below the 3% distortion level, unweighted for both top speeds, except that the test sample's right channel at 3¾ ips was only down 54 dB. These measurements fall within the specs, and are really quite good. Crosstalk was measured at 50 dB down at 400 Hz, both ways, which is just fine for a quarter-track machine. In matters of distortion, or the absence of it, the Viking 433 stayed below 1% harmonic and 2% intermodulation, recorded −10 dB from 0 level. Input and output levels were as specified. One couldn't ask for anything more from the machine.

In sum, the Viking 433 performed quietly and flawlessly in both recording and playing back of tapes. Mechanical operation was positive and smooth, and the sound was definitely "high fidelity." The use of three tape heads, each expressly designed to perform a specific task, enhances performance of the tape deck. The playback head, for example, has a 50 micro-inch gap; the record head, a 500 micro-inch gap; and the erase head, a double gap. All are quarter-track types, naturally.

Thus, it looks like the Viking 433 stereo deck is a winner.

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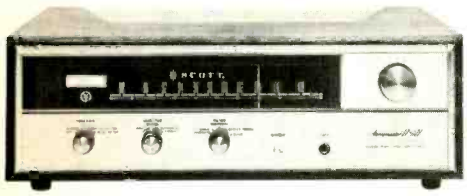
H. H. Scott LT-112B-1 FM Stereo Tuner Kit



MANUFACTURER'S SPECIFICATIONS—Usable Sensitivity (IHF): 1.8 μV. Cross-Modulation Rejection: 90 dB. Signal-to-Noise Ratio: 65 dB. Total Harmonic Distortion: 0.8%. Frequency Response: 50-15,000 Hz ±1 dB. Capture Ratio: 2-5 dB. Selectivity: 45 dB. Stereo Separation (1 kHz): 40 dB. Audio Output Level: 1.2 V rms. Dimensions: 15 in. x 12½ in. x 4⅛ in. high. Price: $199.95.

Years ago the prospective kit builder was warned against attempting construction of a *tuner* until he had attempted less-critical construction projects, such as an amplifier or a preamplifier. A number of important advances have enabled H. H. Scott to develop a tuner *kit* that can be assembled with such ease that even the neophyte kit-builder need not be afraid to tackle it:

(1) Prealigned, modular printed-circuit sections (six, in all) reduce the amount of actual wiring to an absolute minimum. Most of the wiring is confined to interconnections between the modules and the power supply and selector switch sections. (2) The critical front end is completely wired and aligned. (3) Alignment of other tuned circuits can be accomplished quickly and easily without using a single instrument. (4) The 78-page construction manual is written and illustrated in a manner that fairly invites the kit-builder to dive in and "build" without the usual fears associated with kit building. Full color diagrams show actual placement of wires and parts, and assembly and wiring instructions are grouped in easy-to-follow sections. In addition, the manual serves as a well-written primer for anyone interested in the theory of FM and stereo FM.

The kit was assembled by a person who had never made such an attempt before. Except for a slight bending of a dial pointer to eliminate contact with the dial face, and a rather sloppy job of dressing some long wire lengths (which did not affect performance), he did a perfect job in about 10 hours. Precut, pre-stripped and pre-tinned wires assisted greatly.

Features

The finished appearance of the LT-112B rivals that of many quality, factory-assembled units. It does not have a "home-built" look! A rich-looking gold and charcoal-brown dress panel is offset by a subdued dial-glass area that employs soft-blue numerals to indicate frequency, as well as a 0-100 logging scale. The multi-function tuning meter is also contained in the dial glass area. The lower half of the panel contains three selector switches, a stereo indicator light, a standard stereo jack from which an output can be obtained for making tape recordings. (The stereo indicator lamp lights up blue, which doesn't make for good contrast. A red reflector would have been preferable.) The first of the selector switches turns on power to the unit and, in its alternate positions, introduces sub-channel and noise filters for the elimination of noise in less-than-optimum stereo reception situations. The second switch, labelled "selector," enables the listener to choose mono or automatic stereo listening, with or without interstation muting. The third switch selects the various functions of the meter. Its positions include "signal strength" (in which the meter is a "peak reading" one), "multipath" (in which the meter indicates presence or absence of signal reflections detrimental to good stereo FM reception), "center tuning" (in which the meter becomes a "center of channel" indicator, for optimum station tuning) and finally, "align" (in which the sensitivity of the meter circuit is altered to permit its use during initial r.f. and i.f. alignment upon completion of the kit). A close-up view of the meter face is shown in Fig. 2, illustrating its dual scale calibration.

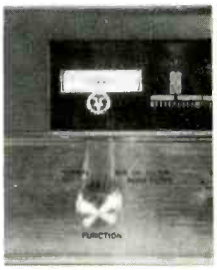


Fig. 2—Closeup of multi-function tuning meter that is also used during final alignment of the Scott LT-112B FM stereo tuner kit.

The tuning knob is located at the upper right of the panel, and its action is fairly smooth and precise, good use having been made of a heavy flywheel assembly.

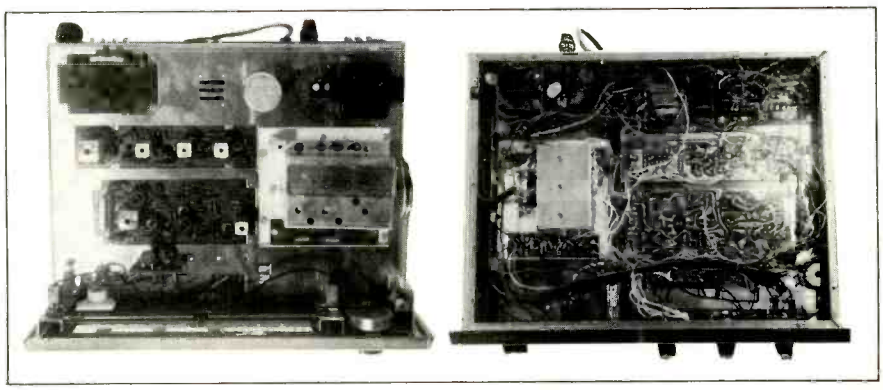
In addition to the usual left and right outputs and antenna terminals, the rear apron of the LT-112B has a pair of jacks for connection to the vertical and horizontal inputs of an oscilloscope. This can provide more meaningful indications of multipath than can be obtained by means of an internal meter. With a 'scope connected, it is also possible to judge centered-tuning, as well as modulation pattern of any given station. For example, Fig. 7 shows a mis-tuned condition with relatively high modulation from an FM station. In Fig. 8 we deliberately modulated a signal generator ±300 kHz (more than would ever be encountered in broadcast practice) in order to display the perfectly symmetrical, wide-band response of the i.f. system. The rear of the tuner also contains right and left level adjustments, so that tuner output levels may be adjusted to match other program source levels associated with the user's overall music system.

Top and bottom views of the completed chassis are shown in Figs. 3 and 4. Careful examination of the underside of the chassis discloses that no effort was made to "dress" wires neatly—we deliberately wanted to check performance of a set that might be built by a rank amateur, as this one was.

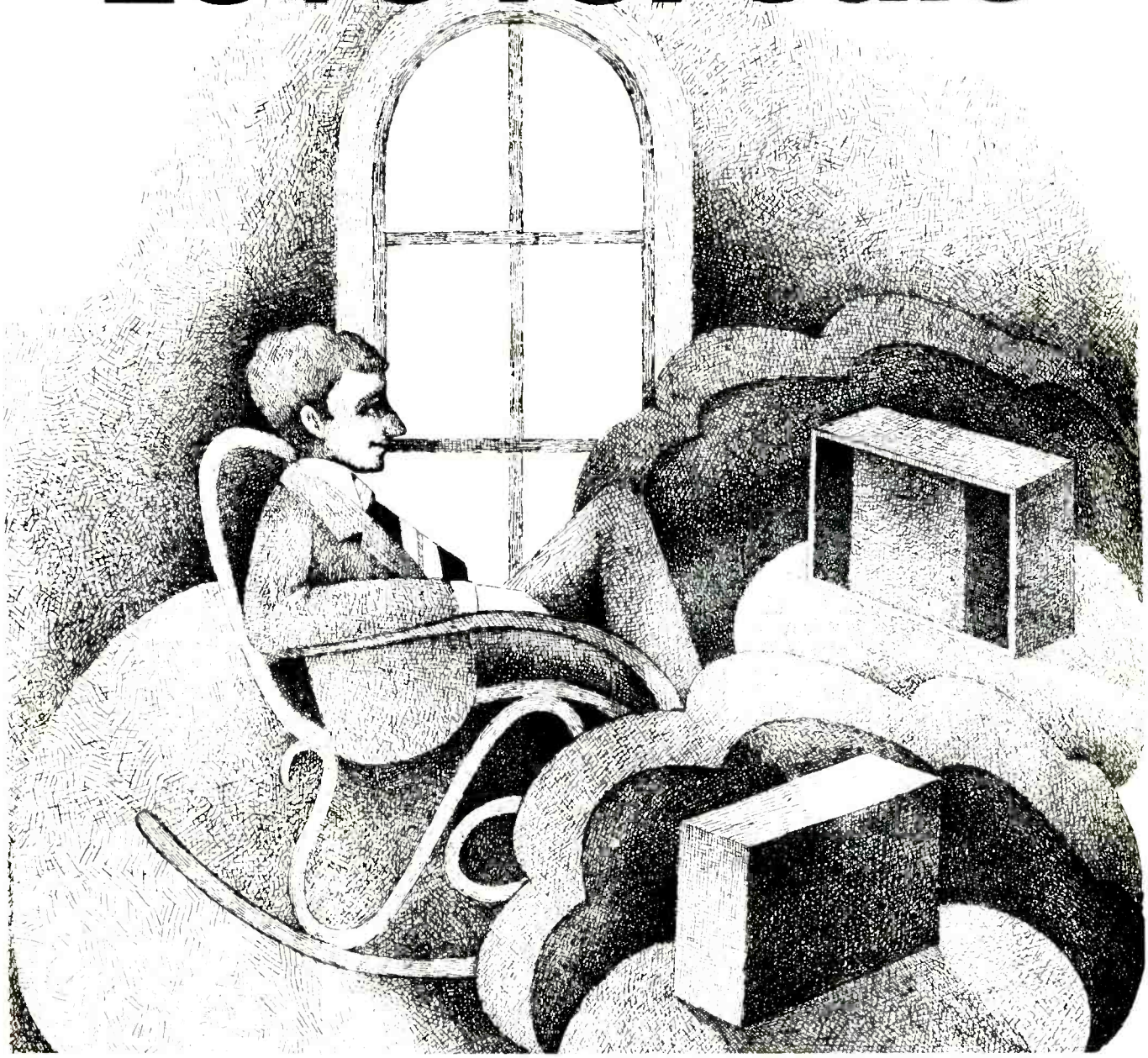
Circuitry

The circuit of the Model LT-112B tuner, as previously mentioned, consists of several modules. The FM front end contains four solid-state amplifying devices, three of which are FET's. Five NPN devices are used in the 10.7-MHz i.f. and limiter strip (which also

Figs. 3 and 4—Top and bottom views of completely assembled kit. Lead wires were not routed as neatly as one would wish, but the tuner operated perfectly after assembly by a kit-assembly "beginner."



Love for sale



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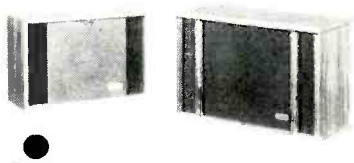

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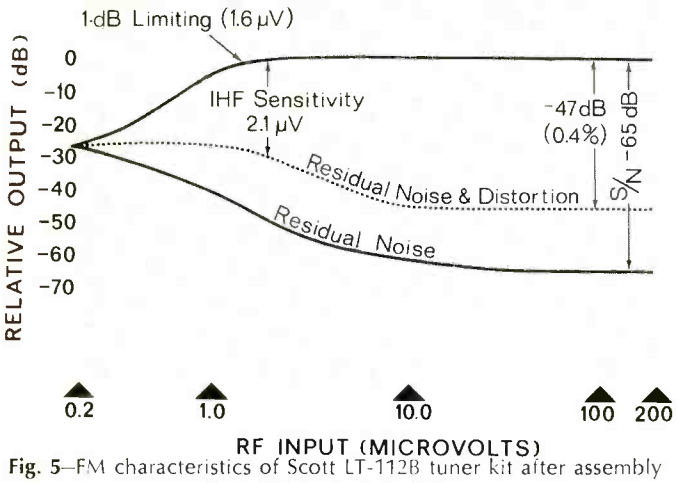


Fig. 5—FM characteristics of Scott LT-112B tuner kit after assembly by a neophyte kit builder.

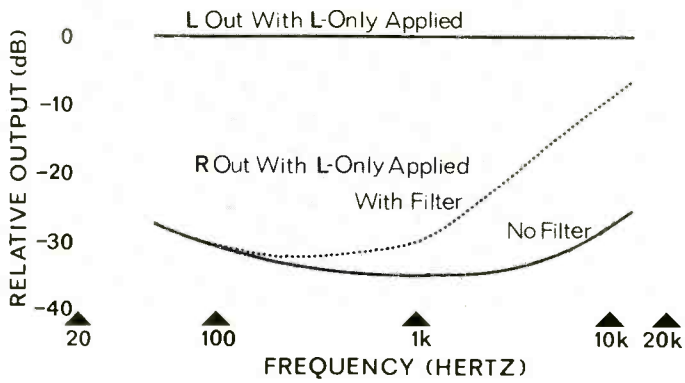


Fig. 6—Stereo FM separation with and without sub-channel filter introduced.

Fig. 7 (left)—With an oscilloscope connected to a pair of jack receptacles provided at the rear of the LT-112B, a mis-tuned condition can be easily detected. Fig. 8—Excellent i.f. bandpass response is displayed with a sweep generator's ±300 kHz applied.

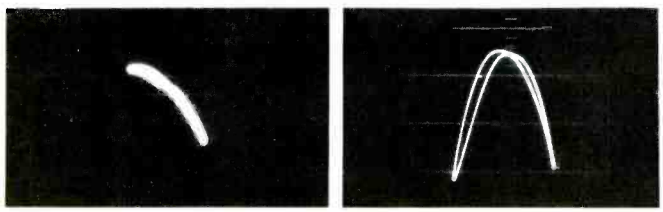
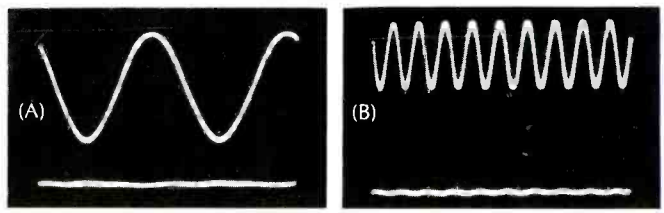


Fig. 9—Visual display of FM stereo separation (graphic plot may be seen above) of the Model LT-112B stereo tuner at (A) 1000 Hz and (B) 10,000 Hz.



employs a ratio detector as the FM demodulator circuit). The multiplex printed-circuit module employs the popular "switching" circuit for demodulation and includes the automatic switching circuits developed by Scott. In these circuits, switching will only occur (to stereo) if there is sufficient pilot signal to insure good synchronization with the locally generated 38-kHz signal. Additionally, the switching circuit requires a greater signal-to-noise ratio for it to switch to stereo than for it to switch back to mono. This prevents a marginally acceptable signal from intermittently switching back and forth from mono to stereo. Seven transistors are used in this carefully designed circuit. A small muting-circuit module consisting of two transistors, a "multipath indicator" module consisting of a transistor and two diodes, and an audio-output amplifier module complete the complement of p.c. boards. The latter includes four more transistors as well as the 38-kHz and 19-kHz rejection filter components and level adjustments for left and right output signals.

Performance

In evaluating the specifications and measurements which follow, the reader is reminded once more that this unit was built per the instruction manual and aligned without the use of any professional test equipment. While it is remarkable that most of the specifications of the completed kit were met or exceeded, further alignment using instruments might, we felt, yield even better results. Amazingly, the IHF sensitivity could not be improved upon using standard alignment procedures. This speaks very well indeed for the factory alignment of the front end, as well as for the techniques developed by Scott for home alignment without the aid of instruments. Much of the FM performance story can be gleaned from Fig. 5. IHF sensitivity at 98 MHz was 2.1 μV, while at 108 MHz (not shown) it measured 1.9 μV, which is right on the specification nose, allowing for normal production tolerances. Ultimate signal-to-noise ratio was exactly 65 dB, as rated, and total harmonic distortion was only 0.4% as against the published figure of 0.8%. Full limiting was achieved with a mere input of 1.6μV!

Stereo FM separation was 35 dB at 1 kHz, a very fine figure, though slightly less than the spec's. Separation at other frequencies is shown in the plot of Fig. 6. Also shown here is the effect on separation when the sub-channel filter is introduced. Note that at the very-high audio frequencies separation is seriously degraded by the filter, as admitted by Scott. Therefore, the filter is really intended for situations in which reduced separation is preferred to very noisy, weak-signal stereo reception. A dual plot of stereo separation is shown in Fig. 9 for 1 kHz and 10 kHz signals.

In use, the Model LT-112B confirmed its measured specifications. At a distance of some 25 miles from the center of the Metropolitan New York area, we were able to receive 38 stations clearly, 12 of which were transmitting stereo FM. None of the twelve required the use of the sub-channel filter. Any evidence of distortion was clearly a case of multi-path problems (as confirmed by the self-contained meter, as well as by scope readings) which were almost completely cleared up by a slight reorientation of our antenna.

If you lean towards kit construction, and have steered clear of FM tuners until now, the Scott LT-112B may well serve as your introduction to this fine program source. You might remove the bottom cover every so often, though, or they'll never believe you built it yourself. More importantly, the stereo FM tuner works beautifully. And there's nothing on the market that is factory-assembled to match its performance, features, and appearance at its price.

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We designed this problem *out* of Grenadiers by facing the woofer *down*.

Instead of bouncing bass waves around the room at random, a Grenadier's woofer reflects them from the floor. They travel almost no distance at all to the primary reflecting surface, and that distance never varies.

Instead of a half-dozen significant reflections that can set up standing waves, you get a single, full-dispersion reflection.

Instead of traversing the width of the room for reflection and reinforcement, a Grenadier's bass tones travel mere inches to the floor for full, faultless, unwavering concert-hall sound.

No standing waves. No Inaudible Woof.

And, naturally, you get the converse benefit, too. No ear-splitting, sonic-boom bass blasts, either.

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Electronic Organs

NORMAN H. CROWHURST

Part 3 of a series

What a home constructor would have to go through to build his own electronic organ, with special focus on generators

MANY READERS of this magazine have probably toyed with the idea of making their own organ. You can do this, of course, by using one of the kits now available from several manufacturers, which can save you much money by doing the assembly yourself.

Schober and Artisan each specialize exclusively in organ kits and, using slightly different approaches, enable you to build as much or as little organ as you like, or can afford. Or you can start small and build on, to make your organ bigger, as funds are available. Heathkit has Thomas organs in kit form, with "Band Boxes" that can be added later to create different rhythms, as well as one to automatically add percussion instrument sounds.

These are ways of saving money and getting a good organ, if you want to do the work yourself. But another idea has intrigued a number of readers: that of completely designing and building their own organ. I know this from the letters I receive asking for advice on how to do one or other part of it.

It seems these readers would like a good organ, but the price tag deters them, and they have convinced themselves they must be able to build one cheaper themselves.

Actually, I'm a great advocate of the home constructor who figures out his own ideas, because many of our best developments started that way. And I know how hard the bug can bite. So maybe you who suggest this should get the bug out of your system by trying it. At least enough to find out what the problems are. After that you will have a healthier respect for the manufacturers who make good organs at what is really a very reasonable price, instead of continuing to think the price is too high!

When you try to build your own generators, the first problem you'll hit is holding the notes on key. Assume you're going to design yourself an oscillator to generate 'A' of 440 Hz. How close does this have to be?

The musical scale is divided into 12 semitones, representing a ratio from one note to the next of 1.059465496:1. This ratio is the 12th root of 2. To sound true, each note must stay close to its own frequency. It cannot possibly wander anywhere near the frequency assigned to the next note.

For precision tuning, musicologists divide each semitone interval into 100 cents, which means this interval is the 1/1200th root of 2, which represents a change in frequency of less than 0.06 of 1%. Actually, so small a change can only be heard by the most highly trained ear.

Musicologists generally concede that most musical persons' ears can detect a change of pitch, as out of tune, when it reaches about 5 cents, which is still only 3/10th of 1% variation in frequency.

Of course, crystal control can hold ratio frequencies much closer than that. But do you intend to crystal control every frequency on your organ? If so, you'll make the organ manufacturers' price look quite low by the time you're through!

To make the many different frequencies produced by an organ stay closely enough in tune, at the cost you have in mind, you're going to have to find a way of making a relatively inexpensive oscillator hold its frequency much better. So let's take a look at the problem.

The Electromechanical Approach

Hammond, and the English Compton, solve it very well by using precision mechanical drives for their electromagnetic and electrostatic generators, respectively. If the whole drive changes speed, the pitch of the organ will change, but it will still be in tune with itself. Do you want to take a crack at copying one of them?

The Compton approach would probably cost you a million dollars, to make just one organ. So that's out for a one-shot deal. Of course, having tooled for the job, you could manufacture organs for much less than that, but they'd never be "cheap."

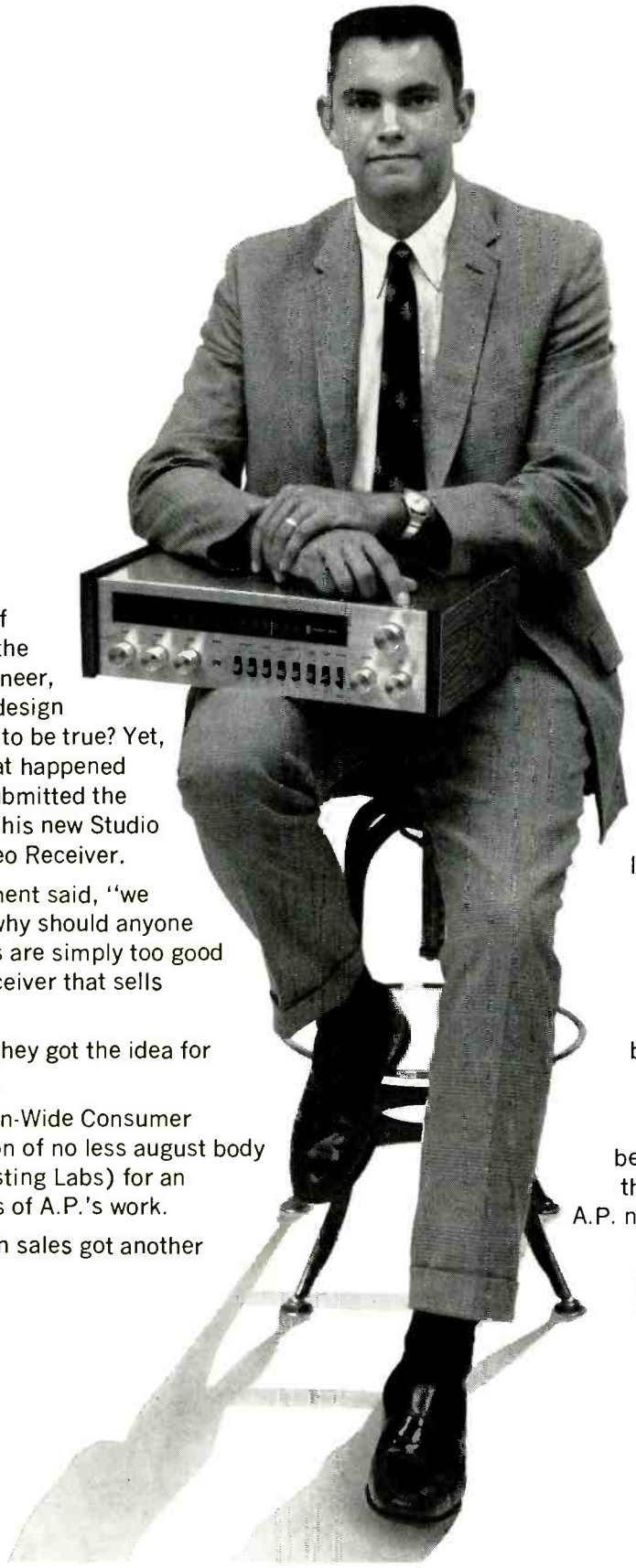
Taking the other choice, how are you going to shape the teeth of the tone wheel, Hammond style? What shape gives precisely sine-wave output? Assuming you are smart at lathe work and can do that part, now you've got to face some pretty complicated switching, between the drawbars and the manual keying, to synthesize the various voices you want.

Hammond has gone through all this and reduced the whole process to a production basis. And they've been producing good organs this way, competitively, for a good many years now. Do you really think you could make a job of it, for less than you can buy a Hammond?

Electronic Generators

So let's look at the possibilities of doing the electronic job ourselves. What kind of oscillator shall we use?

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.

And that's how the University Studio Pro-120 came to be the world's first and only certified receiver. Just because it seemed too good to be true.

What about A.P.?

Well, getting his baby certified made believers out of lots of people. Including the boss. So, instead of a double check, A.P. now has the dignity of a doubled check.

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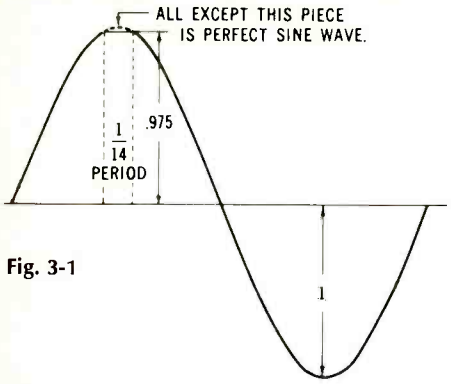

Fig. 3-1

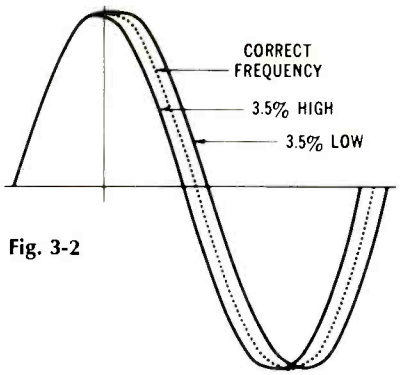

Fig. 3-2

Fig. 3-1—A waveform to illustrate the problem of stability in a tuned-circuit oscillator designed to produce a waveform close to sinusoidal. The waveform is a perfect sine wave, except for a flattening at the top, that reduces amplitude in that direction by 2.5% (0.025). This waveform would measure only 0.15% distortion, but for this reason is susceptible to a frequency deviation of about ±3.5%.

Fig. 3-2—Only the length of the flat part in the waveform of Fig. 3-1 is changed. The rest of the perfect sine wave is unchanged. But the effective frequency change due to this is enough to make the generator unsatisfactory for organ use, regardless of the kind of oscillator circuit used to generate it.

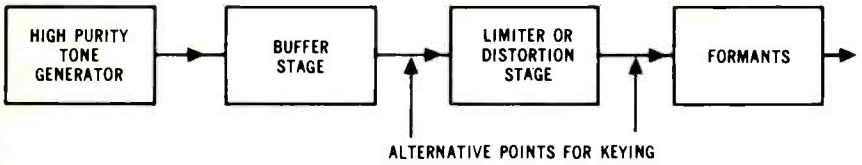

Fig. 3-3—Using the pure-tone concept of tone generator, each note on the organ needs the first three stages shown here in order to use the formant method of timbre selection. Keying can be effected after either the 2nd or 3rd stage. The formants may be common to the whole organ or to whole sections (an octave or more) of notes.

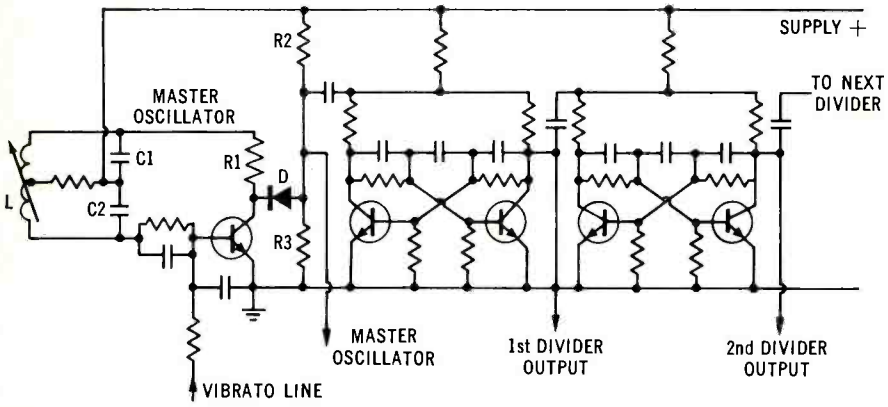

Fig. 3-4—A typical master-oscillator-with-dividers circuit, using transistorized oscillator and dividers. With slight variations, this kind of circuit is used by several manufacturers.

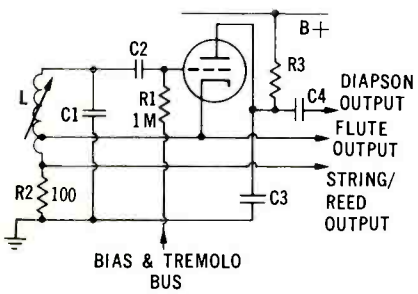

Fig. 3-5—Modern Conn organs use an interesting combination of tubes for oscillators, with transistors for keying. The oscillator circuit, shown here, delivers three different kinds of waveform, used for different groups of stops, which add different formants to these forms.

Will it be a tuned circuit, or a multivibrator type?

In audio engineering, the tuned-circuit type is sometimes regarded as the most stable. To hold stability, the "L" and "C" must stay put. So you buy yourself high-stability, low temperature-coefficient L's and C's. There goes the cost again. By the time you have enough for all the oscillators, you'll have spent enough to buy a pretty good organ, before you even start building.

Actually, for an organ-type oscillator, the variation in L or C value isn't even the biggest cause of frequency change. So you've wasted your money, as I know quite a few people have done. Assume for the moment you have an L and a C that stay absolutely fixed in value. If you can use them in an oscillator that uses exactly 100% feedback, then your L and C will fix the frequency for you.

A variety of classic circuits can make such an oscillator. Here's the rub: if the feedback is 100.001%, the tone slowly builds its output level until distortion stops it from getting any bigger; if the feedback is 99.999%, the tone slowly dies away (assuming you somehow got it started in the first place) and quits.

How are you going to control the gain of your active element (tube or transistor) to give such a precise 100% feedback?

Take the waveform of Fig. 3-1. On an average-reading meter, this has a measured harmonic content of 0.15%, which is pretty pure. The ratio of the peak deviation from sine wave to the fundamental's peak amplitude is 2.5%. And the duration of the deviation is 1/7th of the half wave in which it occurs, or 1/14th of the whole wave.

For 13/14th of its period, the waveform is a perfect sine wave, and thus controlled by the L and C values. For the other 1/14th, it can be influenced by external circuits, via power supply, output coupling, etc. Assuming such external influences can squeeze the actual duration of this nominal 1/14th to half of that, or extend it to one-and-a-half times that, the resulting variation in total period is 1/14th of the nominal period (Fig. 3-2). The only

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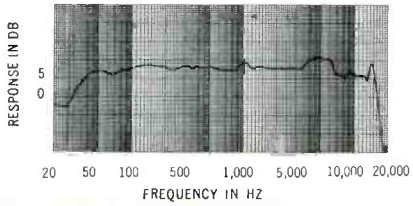

TYPICAL RESPONSE CURVE WHEN MEASURED ON 6 cc COUPLER

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part that changes is the duration of the flat piece.

Thus a frequency adjusted to be 440 Hz can vary as much as from 408.5 Hz to 471.5 Hz. The adjoining semitones have nominal frequencies of 415.3 and 466.16 Hz. So this 440 Hz nominal note can easily wander into the frequencies belonging to adjoining notes. The precision of the L and C values doesn't help at all, unless some very careful isolation is used as well.

But suppose, for the sake of argument, you get a nice pure wave, as viewed on your oscilloscope, and its frequency is stable within, say, 0.3%. This means the circuit has a working Q factor of 300 or more. You can't switch this oscillator on and off as a way of keying the notes. It will have a build-up and decay time-constant of more than 300 periods. Middle C, with a frequency of 261 Hz, will take more than a second to "speak."

So you must keep it oscillating all the time and key its output in some way. How do you connect the output? Any connection to the circuit loads it. To get this degree of purity, you have just the gain to produce 100% feedback, ± 0.03%. This precise amount of feedback depends on a precise resistance value at the part of the circuit where the output is connected. So the external resistance must be more than 3000 times that of the circuit to which it connects if it is not to change the circuit's operation by another 0.03%, with a potential frequency shift of 0.3%. To achieve this, you'll have to use a pretty well-designed buffer stage, and then do your keying after the buffer stage. Already you're thinking of ways of doing the job that will be more expensive than those used by some organ manufacturers.

Actually, most organ manufacturers that use what look like tuned circuits don't even try to maintain that order of purity. The harmonics can be useful as part of the voicing. If you try listening to a sine wave, the first thing that will strike you is that its not much of an organ tone, or any other musical tone. No musical tone ever sounded that "pure." You've got to add harmonics a-plenty to give it any kind of timbre, even flute or diapason. So, having generated a relatively pure tone, you now have a choice: distort it to put in the harmonics, and then use formants to produce various timbres; or synthesize, which means adding in various pure-tone harmonics with switching and separate controls.

The first is relatively easy to do, and controlling timbre with formant circuits is fun, even though it may take you quite a while to find the effects that sound realistic enough to suit you. But it now involves (1) a precision pure-tone oscillator for each note; (2) a buffer stage to isolate the oscillator from the switching; (3) a distorting stage to put in

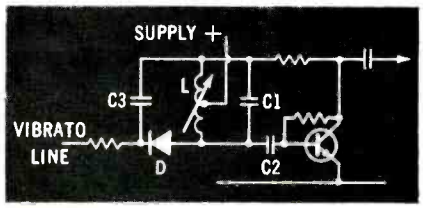

Fig. 3-6—A popular method of getting vibrato. Biasing on the diode, D, parallels small capacitor C_3 with the main tuning capacitor C_1, for part of the vibrato period, thus changing frequency with virtually no amplitude change.

a sackful of harmonics and (4) the formant circuits to produce the desired musical qualities (Fig. 3-3).

As multivibrators produce stage 3 directly, for each note, we can see why most organ makers use a set of master oscillators, to which they couple frequency dividers that work on the multivibrator principle, producing a virtually square wave (Fig. 3-4). Even the master oscillators don't try to work as high purity sine-wave generators. If you've played with the ideas we just finished discussing, you know why!

Actually, it is easier to control the frequency of an oscillator that isn't quite so pure. Although the master oscillator circuit looks like the traditional tuned-circuit oscillator, it is better understood if you view it as controlled by the combined time-constant effects of L, C1 and C2. It oscillates hard, limited by the additional loading of R2 and R3 across the voltage divider formed by R1 and the transistor during the part of the period when D is conducting. Taking the output from this point also obtains a somewhat "squared" waveform. This circuit is similar to that used by Thomas organs.

Using harder oscillation, so frequency is controlled by time constants rather than reactances, yields better stability, but the design still takes a lot of experimental work. Some organs get two or more kinds of tone from the one oscillator (Fig. 3-5). This is one of the Conn circuits. This type of organ uses separate oscillators, one for every note.

The most sinusoidal tone comes from the cathode tap of the oscillator coil. This is used for flute-type tones. The resistor in the ground lead picks up a pulse that reflects grid-current charging on C_2, whose value is smaller than C_1, the main tuning capacitor.

The plate gives an output in which tube current pulses discharge C_3, leaving it to recharge through R_3, yielding a sawtooth output. The inductor has variable-core tuning. This method of providing different basic output waveforms makes voicing with formants easier after keying. Formants can only modify harmonics already in the waveform you start with. They can't completely eliminate any, and they can't add harmonics that aren't there.

A symmetrical waveform has no even harmonics, only odd ones, and thus simulates a closed organ pipe quite well. An asymmetrical waveform has at least a strong second, or octave overtone, which is important for some tone qualities. A sawtooth has them all, fairly richly. And a spiky, or pulse-type waveform, is rich in the higher harmonics, giving a reedy effect.

To provide two or more kinds of output for the whole organ normally requires an oscillator for every note, which Conn uses, not just a master set of 12. As was said in the previous installment, this sounds like "more organ," but its also a lot more work to make and tune.

The synthesis method involves virtually the system used by Hammond, but using electronic instead of electromagnetic sources. Some manufacturers synthesize by combining different waveform outputs, or by using coupling from other generators. But to our knowledge, Hammond is the only one to use complete tone synthesis.

(Continued on page 99)

New developments in the great bass revival.

Last year, when we introduced the Fisher XP-18 four-way speaker system with its huge 18-inch woofer, we predicted a renewed interest in bass among serious audiophiles.

We pointed out that no bookshelf-size speaker, not even the top Fisher models that are famous for their bass, could push the low frequencies around a room with quite the same authority as a big brute like the XP-18.

This came as no surprise to those who remembered that a 40-cycle sound wave is more than 28 feet long. That's why it takes a double bass or a contrabassoon to sound a note that low. Bass and big dimensions go together.

But the sound of the big XP-18 did surprise a lot of people. They knew it had to be good at $330, but they weren't prepared for a completely new experience.

And then came the obvious request: Couldn't we make the XP-18 concept available in more moderately priced speakers?

The new XP-15B

The new XP-12

We could. And did: in the new Fisher XP-12 and XP-15B.

They're a little smaller (24" x 22½" x 13¾" and 27" x 27" x 14¾", respectively), but still twice as big as bookshelf speakers. They're three-way systems instead of four-way, but they have the same type of small cloth-dome tweeter and 8-inch midrange driver with molded rubber surround.

The main difference is in the woofers: a 12-inch unit with a 6-lb. magnet structure in the XP-12 and a 15-inch driver with a 12-lb. magnet structure in the XP-15B.

The prices justify the slight comedown in woof-inches; the XP-12 is listed at $199.95 and the XP-15B at $269.95.

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ABZs of FM

LEONARD FELDMAN

The Discriminator as an FM Detector

Having followed the FM signal through its conversion to an i.f. of 10.7 MHz and its subsequent amplification and limiting, we now have to remove the audio information from its i.f. carrier. This process is known as FM detection or FM demodulation. The two most popular circuits used for this vital function remain basically unchanged. They are the so-called Foster-Seeley Discriminator and the Ratio Detector. We shall first examine the discriminator.

Figure 1 is a schematic of an early form of discriminator and, although it is not in use today, it is simpler to understand than the later-developed Foster-Seeley type. L1 and C1 form the output load of the preceding final limiter stage. This tuned circuit is broad enough to pass the 200 kHz (or more) bandwidth deemed necessary in FM reception. L1C1 energy is inductively coupled to two secondary tuned circuits, L2C2 and L3C3. To obtain FM detection, L2C2 is tuned to a frequency about 100 kHz below the i.f. frequency (10.7 MHz), while L3C3 is tuned above the i.f. center point by an equal number of kHz. Figure 2 is a combined plot of the response curves of the two adjacent resonant circuits. Note that the L3C3 response curve is inverted with respect to the L2C2 curve, indicating an inverted polarity conforming with the actual hook-up and polarities established in Fig. 1. Thus, if the voltage appearing across R1 is larger than the voltage across R2, the net output voltage (with reference to ground) will be positive. A negative resultant output voltage will result if the voltage across R2 is greater than the voltage across R1. It should be noted that each of these resonant circuits may be looked upon as a complete AM detector, including its own diode rectifier, load resistor, and even r.f. bypass capacitors (C4 and C5).

Since each of the resonant circuits is tuned to a different frequency, the amplitude developed across their respective loads will differ, depending upon the instantaneous frequency present. With no modulation present (frequency dormant at 10.7 MHz) equal, small positive and negative voltages will be developed across R1 and R2, respectively. Being opposite in polarity, these voltages will cancel each other out and the resultant will be zero, as it should be for a "no-modulation" condition.

Suppose now that the instantaneous frequency shifts to point "A" as a result of some instantaneous modulation. The voltage across L3C3 will be greater than that across L2C2 because the frequency is closer to the resonant point of the L3C3 circuit. As seen in Fig. 2, the instantaneous resultant voltage developed across the combination load of R1 and R2 will be negative. Furthermore, as the frequency of the carrier (and hence the i.f. stages) shifts back and forth at a rate determined by the audio tone to be reproduced, the output across this combination load will rise and fall through positive and negative values, effectively converting frequency variations into their corresponding amplitude or audio variations.

Fig. 1—An early version of a discriminator (or FM detector).

Fig. 2—Superimposed response curves for the secondary tuned circuits of Fig. 1.

Fig. 3—Combined "S" curve of the discriminator shows linear portion (no curvature) from point 1 to point 2.

Fig. 1

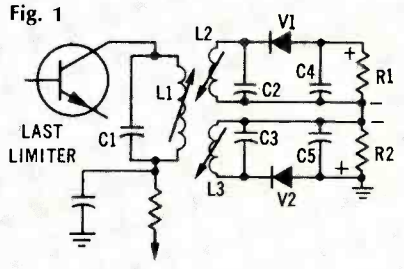


Fig. 2

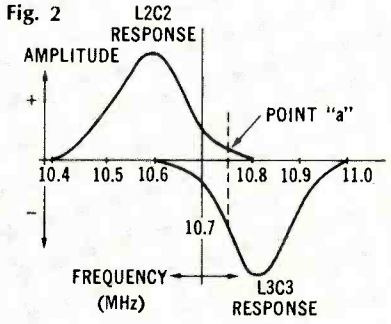
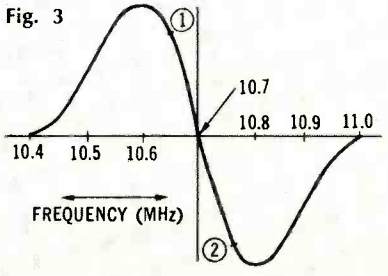


Fig. 3



Since the output voltage is really the difference in voltage across R1 and R2, both curves can be represented as one continuous curve, as shown in Fig. 3. This is the familiar "S" curve so often referred to in alignment instructions for FM sets. The central, linear portion of the curve must be at least 150 kHz from point 1 to point 2 if distortion-free audio demodulation is to take place. Generally, 250 kHz and even more of linear region is designed into these circuits to insure against slight mis-tuning away from center of channel and to further reduce audio distortion.

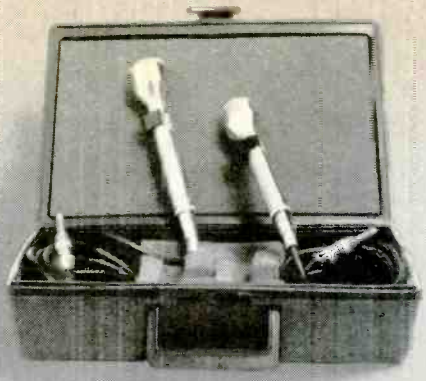
To summarize the action of this early form of discriminator, it may be said that two separate actions occur. First, the tuned sections convert frequency modulation to amplitude modulation at i.f. frequencies. Then, by inserting a diode detector, the audio-modulated i.f. frequencies are converted to the desired audio. From the foregoing, you can deduce that this form of discriminator is sensitive to AM variations, and it is for this reason that limiters must be used ahead of the discriminator, so that the discriminator input is "pure" FM with no AM content.

From the foregoing simple analysis we go on to the Foster-Seeley, shown in one of its many forms in Fig. 4. In this circuit, both secondary windings are combined and a single capacitor is used to tune the circuit to 10.7 MHz. Inductive tuning of both primary and secondary is usually employed. The discriminator output voltage no longer depends upon difference in response of two tuned circuits to various incoming frequencies. Instead, the voltage appearing at each diode will depend upon the *phase* of the secondary voltage as compared to the phase of the primary voltage.

Each different frequency (above and below 10.7 MHz) alters the phase response of the secondary network which, in turn, causes each diode to receive a different amount of voltage. From that point on, however, the action is as described before, in that the rectified voltages across R1 and R2 give the proper audio output.

To illustrate this "phase" of discriminator theory, let us consider what

free



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happens first when the incoming i.f. frequency is at its mid-point (10.7 MHz) or with no modulation applied. The voltage induced in the secondary E_{in} produces an in-phase secondary current, I_s, since, at resonance the impedance presented is purely resistive.

On the vector diagram of Fig. 5, E_{in} and I_s are therefore drawn along the same straight line. The voltage developed in L_2 and L_3 due to I_s is 90 degrees out of phase with I_s. This, of course, is true of any inductance. In the vector diagram, E_2 and E_3 (the voltages developed across L_2 and L_3) are both drawn at 90 degrees from I_s. These two voltage vectors are drawn on opposite sides of I_s because of the reference center tap on the secondary coil. With reference to this center tap, E_2 and E_3 are 180 degrees out of phase with each other. Now, if E_{L4} (the equivalent primary voltage which can be shown to appear across L_4) is added vectorially to E_2 we obtain E_{v1}. By adding E_{L4} to E_3 we also obtain E_{v2}, the two respective voltages applied to the diodes $_{v1}$ and $_{v2}$. It is obvious from the diagram that E_{v1} and E_{v2} are exactly equal in amplitude. Therefore the same current will flow through each diode and similar voltages will appear across R_1 and R_2. Being out of phase, these voltages will cancel and there will be no audio output. Again, this is as it should be, since no modulation is applied to the signal, which remains static at 10.7 MHz.

By way of contrast, let us consider the case in which the FM signal, now modulated, swings towards a higher frequency. As shown in Fig. 6, E_{in} and E_{L4} still bear the same reference relationship to each other (namely, 180 degrees apart in phase). At frequencies above resonance, however, X_L (the inductive reactance) exceeds X_c (capacitive reactance) and the current, I_s, will lag behind E_{in}. E_2 and E_3 still maintain a 90-degree relationship to I_s, however, since that relationship always exists between the voltage and the current in a given coil. If we once again add E_2 to E_{L4} and E_3 to E_{L4}, vectorially, we see that resultant E_{v1} is now greater than resultant E_{v2}. As a result, the voltage developed across R_1 will be greater than that developed across R2, and the output voltage will be positive with respect to ground or the center tap.

A similar, but opposite phase analysis by means of another vector diagram could easily be drawn for the case in which the incoming frequency shifts below center, in which case the output voltage would be negative with respect to ground. The unbalanced condition that arises from the shifting frequency (either negative or positive) is made linear with respect to frequency by careful design of the discriminator transformer so that the audio output will be a faithful replica of the audio which caused the modulation at the transmitter. The S curve previously shown applies equally to this design; the linear portion can be made just as great as in the previous case.

A modification of the Foster-Seeley circuit is shown in Fig. 7. At first glance you might suppose that voltage E_{L4}, the reference voltage needed for proper operation of the discriminator, has been eliminated with the removal of L4. Actually, however, R2 is now effectively in parallel with L1 (thanks to coupling capacitor C3); therefore E_1 appears across R_2. In this circuit, R2 performs a double function—it develops the rectified voltage from the diode and serves to apply E_1, the reference voltage, to the opposite diode. The advantage of this configuration lies only in the fact that fewer parts are required. R1 and R2 must be high in value, however, because they are effectively in parallel with L1. In the original Foster-Seeley circuit (Fig. 4), L4 served as a choke, isolating R1 and R2 from L1. For reasons which we shall go into when we discuss stereo FM, the higher output impedance can sometimes create problems in coupling to a stereo FM decoder.

Whether a discriminator or a ratio detector is used in a given design, there remains one important job to be done before the demodulated audio can be applied to an audio amplifier. You will recall that the frequency response of the program material broadcast is anything but "flat" in the high-fidelity sense. Rather, the high frequencies have been deliberately "boosted" above about 1500 Hz to improve the signal-to-noise ratio of the overall received signal. The. scheme is called "pre-emphasis." In order to restore "flat" response, a de-emphasis network must now be introduced. R3 and C5 in Fig. 7 serve this function. Note that the R-C time constant, as shown, is only 68 microseconds, as opposed to 75 microseconds used at the transmitting end. Usually, length of connecting shielded cable and/or stray wiring capacity make up the difference. Sometimes, less meticulous designers will *under*-de-emphasize recovered audio in order to create a more "brilliant" sounding output, but they are only deluding themselves and the public. Further, this causes inaccurate frequency response and a less-than-optimum signal-to-noise condition.

Next month we shall examine the Ratio Detector.

Fig. 4

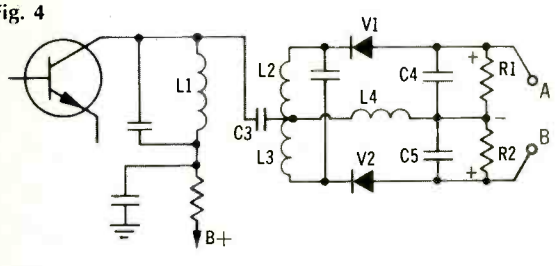


Fig. 4—One version of the Foster Seeley discriminator circuit.

Fig. 5

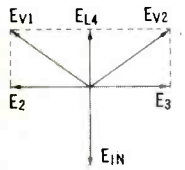


Fig. 5—Vector relationships in the discriminator of Fig. 4 when no demodulation is taking place. Voltages E_{v1} (across R1) and E_{v2} (across R2) cause currents which cancel each other out.

Fig. 6

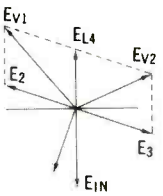


Fig. 6—Phase relationships when frequency is shifted **above** resonance cause E_{v1} to be greater than E_{v2}, and a net output voltage (+) will appear across R1 and R2.

Fig. 7

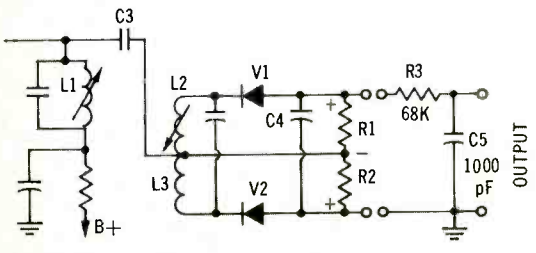
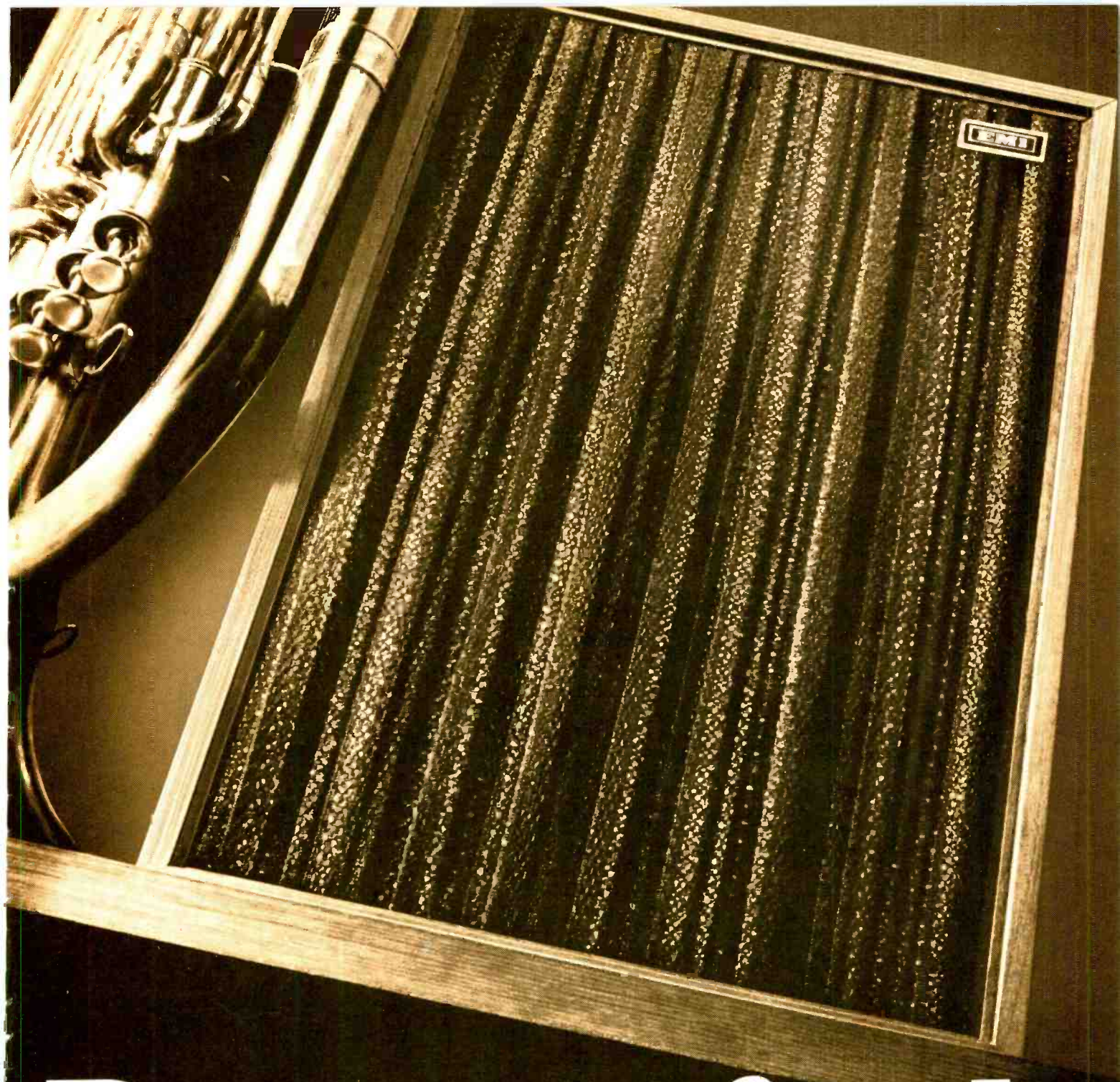


Fig. 7—A modified version of a discriminator, in which L4 and C5 have been omitted, also shows parts needed for proper de-emphasis.



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New EMI 205

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Classical Record Reviews

EDWARD TATNALL CANBY

Four Hands

Though television prospers, we still have radio — broadcasting in totally blind form. For good reason! We don't always want to *see* everything we hear. If silence is sometimes golden, then the absence of visibility is often sweet. Often enough, it is nice just to hear, without seeing a thing.

Music particularly. TV has had a dreadful time, over the years, figuring out what we should look at while serious music is playing in the TV sound channel. (I mean, of course, music to listen to for itself, not background music to the TV foreground.) It's a problem.

In the concert hall, the musicians are situated at a decent distance from you, visible in the round but not so you see the hairs on their faces and the sweat on their brows. On TV, the distant musician is mainly a shapeless blur—not enough lines to the inch, not enough definition. So TV moves in close. Sometimes admirably, as when Toscanini tortured his expressive face before the close-up camera. But more often in a pedestrian, uninteresting fashion; because musicians aren't supposed to act like movie actors when they play. Most of them make meaningless faces, or swoop and sway around, or worst of all, merely look deadpan. Their expression quite rightly is in the music itself. That's where you are supposed to gauge their sentiments, not by looking at their faces.

Also, keep in mind, many mouth-blown instruments require the players to make dreadful contortions. Have you ever looked a bassoonist in the mouth, close-to? Or watched an oboeist nervously licking his reeds, or a trumpeter blowing out cheeks and growing redder and redder in the face? (Bright purple on TV color.) And let us not mention the French horn, who must periodically empty the saliva from his instrument onto the floor. Let's have *that* at a distance, by all means!

To carry this one little step further, we might note—and this is the point of my musings—that even in the "live" form a great deal of music-making does not bear too close a scrutiny via the eyes. In particular, there is that happy home-maker form of musical duet usually called "piano four hands." What you see when you look at it is one piano and two rears. Two people squeezed together on one piano bench, at one keyboard. It can be just lovely for the two participants. But it isn't something to look at from the outside. Thus of all famous and worthwhile forms of classical music, the piano four hands literature is heard the most seldom on the concert stage. A very great loss, since some of the finest classical music around comes in the once highly popular (for home use) format. It is next to impossible to play this splendid music on any stage, of any sort, without extreme clumsiness in the visible aspect.

And lo!—we have records, and stereo too. At last, we can listen to piano four hands for as long as we want without a behind to be seen. And what a boon to the recording artists! They play in privacy as far as prying eyes are concerned. Only the ears hear.

So in due course, over the years, a good part of the four-handed literature for one piano has appeared on records, much of it even before the expansiveness of stereo gave the four simultaneous hands the unmistakable sonic impact that belongs to this special music—twenty fingers that span a keyboard in almost orchestral fashion. The latest album to reach these reviewing ears is typical, the complete works of Mozart for piano four hands, filling up an ample album of three stereo LPs. Some of Mozart's finest music is on these records, the range of expression extending from his earliest childhood years until the last days of his short life. One piece is unfinished; it ends an LP in a startling "cut," right in the middle of an idea. (Easier to manage on records than in the concert hall, come to think of it.)

Yaltah Menuhin and her husband Joel Ryce are the piano team in this recording. The performances were appropriately made in a spacious private music room, sounding reasonably old Vienna though it happens to be in old New York, U.S.A. The two performers integrate their playing to the split microsecond, with never a fraction of a degree of phase shift between the parts; they are one, always. The players do, however, shift places, sometimes one at the top end, sometimes the other. I find that I like the husband on the treble best, but there isn't really much to choose. (You can find which is where via the accompanying notes.)

Sonically, the recording suffers from a minor and common piano ailment, hard, percussive peaks on some notes without a doubt due to reflections and standing waves in the relatively small space—small in relation to a concert hall. Big pianos do that. (Mozart's piano was less powerful, with more in the overtones and a less highly charged fundamental. So—fewer problems in home acoustics.)

Yes, the recording is a budget job, minus studio charges, but the tapes are OK, decidedly. One of these days, somebody's going to issue a stereo recording made on a cassette portable. Then what?

Mozart: The Complete Works for Piano Four Hands. Yaltah Menuhin, Joel Ryce.
Everest 3168/3 Stereo ($14.94)

Performance: B+ *Sound:* B—

More Piano

Hindemith: Sonatas for Viola and Piano. Walter Trampler, Viola, Ronald Turini, Piano.
RCA Victor LSC 3012 Stereo ($5.79)

Hindemith's instrument was the viola and these two sonatas, from 1922 and 1939, represent early and late Hindemith for his own special use as well as for other violists. Of the two sonatas I enjoyed the early one best—mainly because it is one of those early works that doesn't sound like Hindemith! His later idiom is so dogmatically fixed, even if masterfully worked out, that we tend to think—oh no, not *more* of that sound? The early sonata is refreshingly without it.

Of the two types of violist, Hindemith was the scratchy, hoarse-voiced out-of-tune type, a very respectable breed of string performer in musical circles. (Some instruments are allowed that sort of individuality.) But, luckily for us, Walter Trampler is the *other* sort of violist, playing a smooth, creamy viola without a trace of harshness. Much easier in the listening! And the young pianist here, Ronald Turini, is a whiz; he promotes Hindemith's piano parts with convincing energy and enthusiasm. (Note: Trampler also has a companion disc of solo viola music, LSC 2974.)

Performance: B+ *Sound:* B

Prokofieff: Piano Concertos Nos. 3 and 4. John Browning/Boston Symphony/Erich Leinsdorf.
RCA Victor LSC 3019 Stereo ($5.79)

In his younger days, living in Western Europe and often touring America, Prokofieff was mainly a pianist, often playing his own works—to uncomprehending audiences. The four piano concerti thus are bunched together in the twenties and thirties, before his return to Russia. He had a curious piano style, wangingly percussive (in the avant garde manner of that time) and very thickly notated — the piano seldom stops for a moment. There are the now-familiar big Prokofieff melodies here and there; but it was a time when dissonance (for the "classical" modernist) was absolutely the rule, and so you'll have to extract those tunes both from the masses of piano notes, by the millions, and from their rather dissonant harmonizations. In spite of RCA's optimistic heading, "two high-spirited pleasures," the music isn't really *that* easy to take in. Let's not over-do the optimism!

I have the curious impression that young John Browning, the pianist, knows these works a lot better than

the members of the orchestra, which is supposed to have specialized in Prokofieff all these years. There are many surprisingly sloppy bits of orchestral ensemble (surprising, that is, for the Boston Symphony) in the admittedly difficult orchestral backing. Whereas Browning cheerfully whangs forth the trip-hammer piano music with the greatest of aplomb. Could it be, that this orchestral weakness is a symptom of poor rapport between orchestra and conductor? Leinsdorf has since left the Boston, in something less than a state of bliss.

A companion disc, LSC 2897, contains the other two Prokofieff concerti.

Performance: B— *Sound:* B

Daniel Barenboim Conducts Mozart Symphonies (No. 32 in G; No. 35 in D, "Haffner"; No. 38 in D, "Prague." The English Chamber Orchestra.
Angel S-36512 Stereo ($5.79)

Double-threat Daniel Barenboim is a young pianist who is also a young conductor; like many of the new generation, he finds a single musical role too small, and aspires towards the eighteenth century all-around musician type. In not too many more years, most of our conductors will be performers as well, and often composers.

The three Mozart Symphonies, No. 32, not often heard, and the other two very familiar, are given a tasteful and well balanced treatment a bit on the chaste side, as though to lean over backwards from those juicy renditions (with very large orchestra) that used to be standard from the older conductors. I don't mind the chasteness; what seems more questionable is a curious slurring of detail work. In Mozart, the

The Queen of Instruments

EDWARD TATNALL CANBY

One of E. Power Biggs' happiest experiments for Columbia Records, this last year or so, has been his enthusiastic recordings on the pedal harpsichord, a mammoth living-room instrument with two regular finger keyboards in the usual lap-high location plus a complete lower-story mechanism lying underneath, on the floor, a case full of resounding heavyweight strings and a regular organ foot pedal board to operate them. Seated on this instrument's bench, the organist plays with all his hands and feet, exactly as on a pipe organ. And E. Power Biggs, of course, is an organist. So was J. S. Bach. And Bach also owned a pedal harpsichord.

The interesting thing about this now rare and slightly impractical home-type instrument is its very close relationship to the organ of Bach's day. In fact this was its entire *raison d'être*.

There have never been more artfully sonorous organs built than in the Baroque era, as most of us now know by virtue of our own ears and record players. But not matter how big, with no matter how many keyboards—up to four—and dozens of complete stops, every organ had to be pumped up by hand. The organist needed muscular assistance, sometimes with relays of pumpers working on enormous multiple bellows. Under such circumstances, no player could afford to muse along in a revery of improvisation, or compose (like the saintly César Franck) by the hour — while the pumping crew puffed away to keep the pressure up and ahead of the leaks.

The only comparable expert that I can think of in our present world is the air-coupled deep-sea diver whose life depends on a hand pump at the surface. Even he is now out of date.

Moreover, in Bach's day organs, as now, were in churches, but those edifices were unheated in winter, even during the four-hour services that were common on Sundays. With so much cold weather in the year, and human pumpers hard to come by, how could a man like Bach do his vast quantities of musical homework? At home—on the pedal harpsichord.

Thus this somewhat ponderous home instrument (in terms of living-room space) was an organist's dream of convenience. It could be played as an organ, with feet and fingers, but without pumper assistance, in privacy and at leisure. The organist could even "register" his music; for the harpsichord has "stops" like the organ, if fewer and with less variety. Most important, the vital aid of his two feet in addition to ten fingers allowed solo performance of music that was totally impracticable on any other solo instrument but the organ itself—the king of all the instruments then available.

Mr. Biggs is an irrepressible humorist as well as an ever-zestful investigator into organ matters. His first record of the pedal harpsichord (built especially for him by John Challis) was, predictably, the music of Bach. From there, however, he turned to high comedy — booming out such oddities as Chopin and the Schubert "Marche Militaire" with an effect somewhat like a circus steam calliope. That record was quite delightful. I once went so far as to use it for background music in a very zany home movie.

The newest Biggs pedal-harpsichord disc is again serious, and again Bach. Very properly, since Bach is the best known name directly connected with the instrument. The six Trio Sonatas, indeed, are cryptically subtitled "for two keyboards and pedals," which could mean the organ but even more likely indicates the pedal harpsichord that Bach is known to have used.

Nothing, as usual, can be proved. But the sound of the Trio Sonatas, three on each of two Biggs records (plus an extra concerto to round each disc out), is highly convincing on its own. The music is ideal for the massive three-voiced harpsichord effect, the two upper melodic lines each on a finger keyboard and the bass line solidly boomed out on the massive strings of the pedal section. For those who enjoy the big, "walking bass" of so many Bach works, this novel harpsichord sound, far more solid than a mere hand-played instrument, is not to be missed.

It remains merely to say that E. Power Biggs, though occasionally a rather uninspiring player, is capable of rising to new heights when his favorite enthusiasms are involved. His playing of these Trio Sonatas is really first rate, for the most part carefully phrased and the registration on the various hand-played stops and the sonorous strings of the pedal bass unusually imaginative and lively. Don't ever underestimate the Biggs potential. On this counterfoil to the great King of Instruments — which we might properly call the Queen of Instruments—Biggs and Bach make a fine team.

Bach on the Pedal Harpsichord. The Six Trio Sonatas. Vol. I, Nos. 1-3; Concerto in G after Johann Ernst. Vol. II, Nos. 4-6; Concerto No. 2 in A Minor after Vivaldi.
Columbia MS 7124, 7125 ($5.79 ea.)

Performance: A— *Sound:* A—

more crystal-clear is the phrasing and articulation, the better is the musical impact.

Performance: B— *Sound:* B

Three Tchaikowsky Symphonies

Tchaikowsky: Symphony No. 1 ("Winter Dreams"); No. 2 ("Little Russian"); No. 3 ("Polish"). U.S.S.R. Symphony Orch., Yevgeny Svetlanov.
Melodiya Angel SR 40057 stereo ($5.79

One of the best aspects of the LP record is its persuasiveness as a medium for complete collections—like sets of books all bound in the same covers. The three early Tchaikowsky symphonies don't often appear on concert programs; we know the last three very much better. But in LP collector's form they are powerful and useful additions to the basic Tchaikowsky repertory, not only to fill out the complete list of six symphonies but as interesting sources for the later music, throwing much light on it—and, last but hardly least, as worthwhile music in their own right.

The first three symphonies sound like scaled-down models for the last three. They have all the familiar features, but in a lower relief; the "big tunes" are not quite as catchy, the skillful orchestration is no less professional but is put to less effective dramatic use, the noise and bombast are precisely as in the later works but, again, with a more diffuse impact.

Not by much. Tchaikowsky was the very model of a total professional and there is not a note anywhere that is out of place, not a trace of youthful clumsiness. Even the style is already totally Tchaikowsky—you can recognize it in an instant.

Good, authoritative Soviet performances, nicely recorded in very adequate, if slightly old fashioned, stereo.

Performances: B— *Sound:* B

Sibelius Back-to-Back

Sibelius: Symphony No. 2. Antol Dorati/ Stockholm Philharmonic Orch.
RCA Victrola VICS 1318 Stereo ($2.50)

The Sibelius Second was once one of the big Romantic-modern repertory concert pieces, and in many a live symphony concert series it still is. But, alas, present-day musicians have lost the Romantic touch and they often flounder in this sort of music, unless they have a "maestro," young or old, who can teach them how to play it—the way Beecham, Stokowsky, Kousse-

vitsky made the music sound within the memory of plenty of us.

Not Dorati. His is an accurate performance via a good orchestra; nothing is smeared, blurred; there are no sour notes. But the spirit of the music, the intention of the superb orchestral effects, is incredibly faulted. The music is parrotlike, point after point missed cold. It is as though one were reading Finnish out loud without understanding a word. Such an expressive and dramatic idiom, too, even if dismally old fashioned, and so marvelously written for playing! I was appalled and fascinated.

If you are an old Sibelius fan you will be much disappointed. If you are new to that skillful composer of big Romance, this recording is a poor way to get to know him.

Performance: C— *Sound:* B

Sibelius: Symphony No. 2. Sinfonia of London, Tauno Hannikainen.
Crossroads 22 16 0026 Stereo ($2.49)

Yes, it's conducted by a genuine Finn, out of Sibelius-land; and the English (the orchestra) have always been the foremost champions of Sibelius. But I found this performance correct and dull. It is accurate enough but somebody — the conductor — just doesn't understand what it is all about, misses the grand tensions, the climaxes, the passages that are supposed to be supremely distraught, or grandly triumphant. Definitely, I wouldn't want to recommend this as a model for those who are curious as to what makes Sibelius tick.

Performance: C+ *Sound:* B

Ormandy/Bernstein in Space

2001—A Space Odyssey; Aniara—An Epic of Space Flight in 2038 A.D. Ormandy Philadelphia Orchestra, Bernstein N. Y. Philharmonic, Gregg Smith Singers, Added Electronic Effects by Morton Subotnik.
Columbia MS 7176 Stereo ($5.79)

Yeh, yeh, we know all about it. Tie-in publicity and all that. Frankly, I found this record a pain in the aural neck, and not because the individual segments are necessarily "bad." It is just too, too contrived, for cash-in purposes.

You've probably seen "2001" by this time. If not, you'll find when you do that the electronic music (Ligeti) and the several "classical" items, such as "Zarathustra" of Richard Strauss and "Blue Danube" by Johann S., blend remarkably well into the unusual picture with its stunning combination of space-spectacle and everyday normality. But to hear these items excised and separate from the film is another matter, especially in the uncomfortable tape-editing synthesis here displayed. They've simply taken the war-horses out of the Columbia catalogue and spliced 'em together, along with the electronic (and part-live) Ligeti items, the whole smeared up with an overlay of electronic bridge music by Subotnik. Ghastly, I say. But you'll maybe like it as trick background Musak.

As for the Blomdahl "Aniara," a self-styled space opera, that also is a bit of happy publicity coincidence. (Imagine it, only 37 years apart!) The Blomdahl was out a good while ago on records and I found it pretentiously old-fashioned in spite of its super-modern exterior, all electronic hoopla of the sort now too familiar right and left. So you can have *that* side as well, for your Musak.

(But do see the movie — and also, afterwards, go out and buy the tie-in book, which oddly enough goes a long way to explain the background of the confusing and controversial last portion of the film.)

Electronics and Percussion—Five Realizations by Max Neuhaus.
Columbia MS 7139 Stereo ($5.79)

No matter how far-out, avant garde art these days tends to fall as rigidly into stylized schools as far-out politics. In five seconds you will know that this record belongs to the special school that goes in for fragmented blips, plops, oozes of sound, minus beat and minus "tunes," much of the material being pregnant silence. Decidedly a school of thinking, though many composers belong to it, each in his own way. This is also the school of the free-form "score" —which merely lays down options for performance that assure no two versions of ever being the same. Like, for instance, the score here which is on a single page which may be read "with any edge up," or another score that is circular and can begin at any point, moving either clockwise or counterclockwise back to the beginning. Still another "score," by that old-time shocker John Cage, is realized here via mikes set in front of loudspeakers to produce random (but controlled, within the set limits) squawks and howls of electrical feedback! Ouch.

Lots of stimulating ideas here and a vast quantity of plicks and plops and gurgles, worth listening to if only as an exercise for the ear in precision sonics What else is any music, after all?

Performances: ? *Sound:* B

Benny Goodman/Bartók

Bartók: Excerpts from Microkosmos; Contrasts. Bartók, Szigéti, Goodman (1940).
Odyssey 32 16 0220 Mono ($2.49)

Bela Bartók himself plays some of his many "Microcosmos" piano pieces on side 1; Bartók, Benny Goodman and Joseph Szigéti play the Goodman-commissioned "Contrasts" for piano, clarinet and violin, on Side 2, these being reissues of notable 1940 78-rpm recordings—I own the original "Contrasts" album, and once reviewed it.

For pianists, especially, the Bartók piano renderings are good listening and instructive. Nothing very difficult for today's ears, and lots of Hungarian-Bulgarian folk tunes to help. "Contrasts" is a wry, dry, astringent piece with an ever so delicate sense of irony, a dusty feeling of jazz, that fits Benny Goodman's clarinet perfectly. Very dissonant, but very expressive too. The recording is clean enough, but lacks highs. The clarinet comes through best.

Performances: A *Sound:* C+

Far East . . . and West

Messiaen: Turangalia Symphony.
Takemitsu: November Steps. Toronto Symphony, Ozawa.
RCA Victor LSC 7051 (2) stereo $11.58)

Messiaen's enormous late-Romantic symphony in modern idiom dates from the 1940s and is the biggest, most blatant sound-blast on a very high level) I've listened to for many a month. For all its skillful complexity and its extreme dissonance, for all the whoops and wails of the Ondes Martinot (a French electronic instrument), for all its immensity of physical size and of length—ten movements on three LP sides — I found it overblown and, in places, just plain corny (in a high level way). It hasn't a tenth the subtlety of Mahler's big music, nor any of the originality. It goes very well with such as the much-vaunted Penderecki, or the more monumental pieces by Carl Orff. But it makes for marvelous hi fi, I'll have to admit.

By contrast, the spare, sparce "November Steps," by a Japanese, combining two ancient traditional Japanese instruments with a "Western" instrumental ensemble, is quite beautiful. The flute tones and the twang sounds are of terrible intensity, depending much on silence, on sudden bursts of ferocity. The orchestral dissonant background is used only a few times, and blends perfectly. Terrific, particularly for young-

er listeners who like the new Oriental slant to our music.

Performances: A *Sound:* A—

Flower Dance. Japanese Folk Melodies. The Noday family and others.
Nonesuch H-72020 stereo ($2.50)

One can never be quite sure how to judge the music of another culture. I can only say that, inside a charming cover (a line drawing adapted from the ex-Checkmate stylings) there was music that struck my Western ear as sort of uninteresting—but how wrong I could be. The music is all-instrumental, consisting of plucked-instrument sounds and the sounds of flute-like instruments. It has, to be sure, a fascinating monotony—it is all the near-key of G minor, or the modal Japanese equivalent. There is near-harmony, but you'll hear no outright chordal passages; the music is not corrupted (as some is) by Western influences and TV-style harmonizations.

Excellent hi-fi close-up recordings, fit for the fanciest listening systems. The record is part of Nonesuch's wide-ranging Explorer Series.

Performance: *Sound:* A—

Chinese Classical Masterpieces. "Youth" (Concerto for Piano and Orch.). Liu Shih-Kun, Pf. **"The Butterfly Lovers" (Concerto for Violin and Orch.).** Shen Yung, Vl., Chinese Conservatory Orch., Fan Cheng-Wu.
Everest 3212 Electronic Stereo ($4.98)

A fancy title but a very sad content. As the notes say, this is music "for the masses" (and it seems to emanate from China, though just how I cannot imagine). If so, then the Chinese masses are getting not Chinese pap but watered-down Western. The stuff is perfectly awful, full of a pathetic pseudo - Hollywood pretentiousness, aping the very worst of sleazy Western music without even the beginning of an understanding—even of our worst.

Yet — the instrumentalists, soloists and orchestra, are highly competent. They play like Western pros! Just goes to prove that you can always export a muscular artistic skill, but the content that goes with it is something else again.

Keep in mind, incidentally, that the Russians first imported Western music as far back as the sixteenth century and have been acclimatizing it ever since. Not so the Chinese. It's new for them.

Performances: B *Sound:* B—

Jazz & Blues

BERTRAM STANLEIGH

Wes Montgomery Legacy

The Best of Wes Montgomery, Vol. 2.
Verve Stereo V6-8757

When one has recorded as much as Wes Montgomery did in the mere nine years that he was a national figure on the jazz scene, it's a bit hard to crowd all of an artist's major achievements onto a single album. Recognizing that volume 1 omitted a number of likely "best" candidates, Verve has repackaged a second set of numbers from earlier collections. This time including *Bumpin'* and *California Dreamin'*, both title tunes from sets that date back a year or two. The only problem with this reshuffling technique is that it probably won't stop until all of the old Montgomery material has been reissued in every possible combination. But aside from the obvious commercial considerations, this set does contain some of Montgomery's strongest late work, with large instrumental backgrounds. And the sound is simply splendid.

Performance: A *Sound:* A

Wes Montgomery Trio: 'Round Midnight.
Riverside Stereo RS-3014

A reissue of the original Riverside first recordings of Montgomery, now being distributed by ABC Records, this set provides a fascinating contrast to the Verve reissues noted above. Melvin Rhyne, organ, and Paul Parker, drums, function as close collaborators, playing real jazz. This is very different from the slickly adroit string backgrounds by Claus Ogerman, Oliver Nelson, and others on his later, more commercial, releases, but Montgomery was one of those rare jazz musicians who produce exciting, meaningful musical experiences no matter what kind of aggregation or repertory may be involved. This set offers a mixture of pop standards: *Yesterdays* and *Too Late Now*, Montgomery's own *Missile Blues* and *Jingles*, and Monk's *'Round Midnight*, Ellington's *Satin Doll*, Horace Silver's *Ecorah*, and Benny Golson's *Whisper Not*. Sound and stereo are both quite satisfactory without, of course, beginning to approach the brilliance of the newer sets.

Performance: A *Sound:* A—

Jazz "Greats" on Riverside

When the first reissues of the Riverside label were released under the auspices of ABC Records, it was evident that they had been culled from that portion of the Riverside catalog that was primarily of interest to mainstream and modern jazz fanciers. Bill Evans, Thelonious Monk, Wes Montgomery, Johnny Lytle, Sonny Rollins, and Yusef Lateef were among those performers whose important contributions to Riverside were happily re-instated. A second ABC release of treasures from this same source includes additional Monk, Evans, Montgomery, and Lytle, and it gives us, as well, Art Blakey, Max Roach, Tad Dameron, Milt Jackson, and the less familiar recordings of Joe Albany and George Russell.

Missing from the ABC-Riverside reissues are all of the early Riverside transfers from 78s of classic jazz that originally graced the Gennett catalog. Now these are also coming out in new transfers on a new label, Orpheum. For its first release, on its "Jazz Legend Series," Orpheum has issued five albums "rechanneled for stereo" that feature the New Orleans Rhythm Kings, Bix Beiderbecke, Jelly Roll Morton, Muggsy Spanier and King Oliver's Creole Jazz Band. The later item has kept its somewhat deceptive title *The Great Louis Armstrong, 1923*. Louis, as every student of jazz history knows, had joined the Oliver group for its legendary season at Chicago's Lincoln Gardens, and his first records were cut in 1923 while he was playing second trumpet to the great King Oliver.

All of the material on these five albums was originally recorded by the Starr Piano Company, which made Gennett records. With the exception of two numbers recorded in New York in 1924 by Beiderbecke and George Brunis, all of the originals were waxed in Richmond, Indiana, between 1922 and 1928. These were not modern electrical recordings. Indeed they were not even the best of acoustical recordings, and it is plain that these new reissues, even "rechannelled for stereo," do not represent any startling improvement in sound quality over the Riverside LPs of the late Forties and early Fifties. They fill a gap in our historical archives, and their reappearance is most welcome, but I have reservations as to whether these particular reissues are as good as they might be. It is no more than suspicion on my part, but I have a hunch that these new masters were cut from the original 78-to-tape transfers made by Riverside some years back. A lot has been learned since that time about how to get the most from an old 78.

Even if there may be more on these platters than can be heard in the present transfers, there is still a lot of good listening to justify their return to the catalog. While some of the material was duplicated by other early record companies, the Muggsy Spanier and Bix Beiderbecke sets offer rare early performances by these two trumpeters that do much to place their careers and later achievements in proper perspective. The Jelly Roll Morton piano solos are also documents of value, and unlike some of the other sets in this series, the Morton sessions of 1923 and 1924, at which this dozen numbers were recorded, have been properly referenced, even though details of the original discs are omitted. If Orpheum has plans to bring out more of the early Riverside cache, we hope they will make an effort to achieve a brighter, closer sound and present more complete documentation. There is probably no chance that they could be cajoled into leaving these mono recordings in their original single-channel perspective. Orpheum's "rechanneling" has been done without resorting to very extreme devices. As a result, there is a slight difference in balance between right and left channels when heard alone, but the final blending of the two channels results in nothing but a slightly tubby mono sound.

The Great Muggsy Spanier
Orpheum Stereo 101

Performance: A *Sound:* C

The Great New Orleans Rhythm Kings
Orpheum Stereo 102

Performance: A *Sound:* C

The Great Jelly Roll Morton
Orpheum Stereo 103

Performance: A+ *Sound:* B

The Great Bix Beiderbecke
Orpheum Stereo 104

Performance: A *Sound:* C

The Great Louis Armstrong, 1923
Orpheum Stereo 105

Performance: A *Sound:* C

Jazz a la Organ

Luiz Henrique and Walter Wanderly: Popcorn.
Verve Stereo V6-8734

As a delightful set of Bossa-jazz or as a stereo spectacular, this set is a knockout. Without resorting to ear-blasting volume levels, Messrs. Henrique, guitar and voice, and Wanderly, organ, abetted by flute and percussion, turn in some of the crispest, close up, musically natural sound anyone could want in his living room. Frankly, I'm happiest when Henrique does his singing in Portuguese rather than English, and I would have been grateful if *Home on the Range* hadn't turned up smack in the middle of side 2, but there's so much brisk, up-spirited music making here that such reservations really amount to quibbling.

Performance: B+ *Sound:* A+

Dick Hyman and "The Group": Sweet Sweet Soul
Command Stereo RS 933 SD ($5.79)

Playing organ and clavinette, Dick Hyman whips up a bit of a storm with the able assistance of Bobby Rosengarden's percussion and Bob Haggart's bass. This is richly melodic jazz that's exciting both for the technical prowess of the performers and the clear, bright sound of the recording. It's a rather small aggregation for a Command record, but it has as much big sound on it as if it held a full orchestra.

Performance: B+ *Sound:* A

Miscellany

T-Bone Walker: Funky Town. **Blues Way Stereo BLS-6014** ($4.79)

A guitarist-singer with a strong, rocking drive and a deeply felt personal message, Walker is heard in a collection of his own compositions with a band that wastes little time on subtlety or low level passages. Walker's fine guitar solos are over-amplified, as is the electric organ, and the reeds in the band rarely drop below a *fff* shriek. There is much fine music-making on this record, and the engineering is not bad, but the combination of continuously loud performance levels and close-up miking make this release a bit painful for extended listening.

Performance: A— *Sound:* B

David Newman: House of David. **Atlantic Mono 14899** ($4.79)

Tenor saxophonist David Newman was a leading light of the Ray Charles band in the early sixties. After a departure from the jazz scene for three-and-a-half years, he returns to discs with a solo set of attractive mainstream statements. With the sure support of Koussie Gardner, organ, Tod Dunbar, guitar, and Milt Turner, drums, Newman makes clear that his return is a welcome one.

Performance: A *Sound:* A

Light Listening

STUART TRIFF

Music from "Man of La Mancha." Joe Sherman and The Washington Squares.
Columbia CS-9657 ($4.79)

Though it's not likely to suit all tastes, this is a decidedly fresh and novel approach to an instrumental version of a Broadway show score. Arranger Sherman has blended Spanish rhythms and Dixieland to give Mitch Leigh's music a new dimension; it works beautifully and the renditions are truly exciting, from the title tune to the popular "The Impossible Dream." Clean, sharp stereo definition.

Performance: B+ *Sound:* B+

Strings Latino: Edmundo Ros and His Orchestra.
London Phase-4 Stereo SP-44107 ($5.79)

A mixture of old and new latin tunes in the typical Edmundo Ros manner, meaning an album of pleasant listening, albeit a bit dated (echoes of Xavier Cugat in his heyday). Nothing old-fashioned about the reproduction, though, being quite up to the standard we've come to expect of Phase-4. The tunes include such Ros specialties as "Rhumba Rhapsody," "Green Eyes," and "Delicado," with the old maestro supplying the vocals on "A Man and a Woman" and "Thank U Very Much."

Performance: B *Sound:* B+

"Angel of the Morning" and Other Hit Themes for Young Lovers. Percy Faith Orchestra and Chorus.
Columbia CS-9706 ($4.79)

Another Percy Faith offering resulting from his apparently successful formula of "in" theme albums, dedicated to the Younger Set. This is a collection of current favorites in warm, intimate renderings by silky Faith strings, augmented by an all-femme vocal group. Included are "Scarborough Fair/Canticle" and the "Mrs. Robinson" theme, both from the movie, "The Graduate"; and two Burt Bacharach tunes—"Do You Know the Way to San Jose?" and "This Guy's in Love with You." The standout cut is a really beautiful instrumental version of the talented Jim Webb's "MacArthur Park." Top-notch stereo reproduction.

Performance: B+ *Sound:* A

Music from "The Young Girls of Rochefort." Composed and Conducted by Michel Legrand.
United Artists UAL-3662/UAS-6662 ($4.79)

As is usually the fate of most sequels, "The Young Girls of Rochefort" suffers by comparison with Legrand's score for "The Umbrellas of Cherbourg." The former, however, has much to recommend it, and the composer's musical creativity and melodic invention are still head and shoulders above the output of most of today's tunesmiths.

Unless you share Legrand's excessive fondness for vocal writing in the instrumental "do-be-do" technique, you'll find this orchestral version of the score more enjoyable than the original soundtrack recording. The arrangements here are, of course, by Legrand, and particularly fetching is the "Theme du Concerto," a tongue-in-cheek opus inspired by the stereotyped Hollywood cinema concerto. Bright and effective stereo.

Performance: B+ *Sound:* B+

Don Shirley in Concert. Don Shirley Trio.
Columbia CS-9684 ($4.79)

As an old Don Shirley fan, I've often lamented this superb pianist's relative inactivity in the recording studios. Now, Columbia has had the good sense to make a "live" recording of his recent Carnegie Hall concert and the result is in every way a notable event. Sensitively supported by the outstanding work of cellist Gilberto Munguia and bassist Henry Gonzalez, Shirley applies his classical, chamber music approach to an artfully-selected group of popular standards.

The program includes a seven-minute sonata version of Vernon Duke's "I Can't Get Started" (with some exquisite cello passages), and a scherzo-like "I Feel Pretty," lasting little more than a minute, plus lovely tone poem treatments of "Yesterday," "I Cover the Waterfront," "By Myself," and the touching "Water Boy." Very commendable sound for an "on-location" job. Run — don't walk, friends, and listen to a magnificent artist in one of the year's outstanding releases!

Performance: A+ *Sound:* B+

Jack Jones: If You Ever Leave Me and ten other songs. Arr. & cond. by Marty Paich.
RCA/Victor LPM/LSP-3969 ($4.79)

Easy listening all the way as Jack Jones swings gently through his newest collection of songs, mostly of recent vintage, with a couple of oldies added for good measure. He brings an ingratiating warmth and intimacy to such tunes as "Goin' Out Of My Head," "By the Time I Get To Phoenix," "The Letter," and "Baby, Don't You Quit Now."

Also included, is "There Comes a Time," from a musical that blitzed in London not long ago, called "The Four Musketeers." Fine backings by Marty Paich.

Performance: B+ *Sound:* B+

Discovery: Larry Adler, harmonica; with Morton Gould and his Orchestra (George Gershwin's "Lullabye Time" plus music by Porter, Gould, Rodgers, Kern, Arlen, & Gershwin). **RCA Victor LM/LSC-2986** ($5.79)

For his recording debut on RCA, harmonica virtuoso Larry Adler has collaborated with conductor-arranger Morton Gould in this album of esoteric material, garnered from the musical trunks of some of the elite among American show composers. When musical compositions remain in the form of unpublished manuscripts and are relegated to a dusty attic by their creators, it is usually for a good reason . . . a reason which this collection (with one exception) makes all too clear.

Fortunately, the Adler-Gould explorations have uncovered one fascinating item that makes this disc worth having—the 8-minute Gershwin work called "Lullabye Time." The version heard here is Morton Gould's tasteful and wholly appropriate transcription for harmonica (playing the first violin part) and string orchestra. The performance is beautiful; played with sensitivity, warmth, and obvious affection.

It's a pity to report that after "Lullabye Time," this collection goes far astray musically and artistically. The other numbers, though pleasant enough, and perhaps even a few notches above most of the pap that passes for popular music today, are really not distinguished.

Performance: A ("Lullabye Time")
C (Everything else)
Sound: B+

Recorded Tape Reviews

BERT WHYTE

Bernstein/Watts: Brahms Piano Concerto

Brahms: Piano Concerto #2 in B Flat. Andre Watts, piano. Leonard Bernstein cond. the New York Philharmonic Orch.

Columbia MQ999, **4 tr., open reel, 7½ ips** ($9.95)

Maybe you have noticed that there have been quite a few recordings of the Brahms B-flat concerto issued lately. No one seems to know why this sudden interest . . . there isn't any kind of anniversary or Brahms festival in sight. In any case, this reading by Watts and Bernstein is one of the latest and, from many aspects, one of the most attractive.

For one thing, young Andre Watts is possessed of a phenomenal technique. The difficult runs, octave progressions, trills, he tosses off with almost insouciant ease. Yet for all his Horowitz-like proficiency, he is an extraordinarily communicative musician with a maturity that has come at a very early age. His is a bravura reading, intensely passionate and poetic. Yet his traversal of the third movement was exceptional for his handling of texture, his clean phrasing, the warmth, and depth of his involvment, the utter elegance of his playing. In this concerto, where the orchestral part has considerable weight, Bernstein's accompaniment (while generally sympathetic) is almost too intense. He allows his sense of drama full sway and, at times, comes close to overwhelming Watts. If you can walk this kind of ragged edge . . . indulge in this "musical brinkmanship," the results can be very exciting.

As to quality, this is a recording notable for superb balance. Balance between piano and orchestra, and between orchestral choirs. Left/right directional effects were nicely proportioned with the piano solidly in the phantom center channel. Frequency response wide with a good solid bass. Dynamics were wide, transient response excellent, as attested by the bright, clean sound of the piano.

This is another of those recordings which must be played at a fairly high level for best balance. At this kind of level, the hiss was somewhat obtrusive. There was also some print-through and some occasional crosstalk. On an overall basis this recording is not as sonically spectacular as some others I have heard . . . and I have yet to audition several other new recordings of this concerto. Nevertheless, with Andre Watts' altogether ingratiating performance and the thoroughly respectable sound, this must be reckoned as a solidly competitive recording that will appeal to many people.

The Versatile Von Karajan

Liszt: Les Preludes; Hungarian Rhapsody No. 2. Smetana: Vysehrad; Die Moldau. Herbert Von Karajan Cond. the Berlin Philharmonic.

Ampex/D.G.G. DGC9037, open reel, 4 tr., 7½ ips ($7.95)

This man Von Karajan is amazing in his versatility. What a study in contrasts is represented on this tape . . . the rather pompous and overblown "Les Preludes" gets a performance of such majestic power as to make the

(Continued on page 96)

piece newly palatable. The 2nd "Hungarian Rhapsody" is also powerfully conceived, but the rhythmic aspects are not neglected. "Vysehrad" is part of Smetana's epic cycle, "Ma Vlast," as is the much more familiar "Die Moldau." Both are given a beautifully atmospheric and evocative reading, featuring superb string work from the orchestra.

There seems little doubt that the combination of Von Karajan and the Berlin Philharmonic has become something of a phenomenon. Certainly he is one of the most recorded of today's conductors, probably because of his aforementioned versatility as well as for his considerable personal magnetism and "box office" appeal. His income from recordings must be tremendous . . . he has so many recordings currently available I could fill this entire column with them! The sound he gets from Deutsche Grammophon doesn't hurt either. It is rarely less than first class and, in many cases, as in this recording, quite outstanding. Beautifully balanced and proportioned, wide in dynamics, appropriate in acoustic perspective, possessed of all the stereo virtues, this is a winner in every respect including top notch EX-Plus processing.

Shostakovich: Symphony No. 10 in E Minor. Herbert Von Karajan Cond. the Berlin Philharmonic.
Ampex/D.G.G. DGC9020, open reel, 4 tr., 7½ ips ($7.95)

Shostakovich's 10th Symphony is slowly becoming recognized as one of his major works. It doesn't assimilate too easily the first time around, but subsequent auditions help to sort out its complexities and you begin to realize what a powerful personal statement the composer has made.

It would seem that this is also one of those "conductor's" symphonies, an ideal vehicle for the aristocrats of the baton. Hence there have been three recordings issued recently of which this is in my opinion, by all odds the best. It is obvious Von Karajan is enjoying himself here, abetted by his magnificently responsive Berlin Philharmonic. This is a very dynamic performance, very intense and dramatic, a supercharged reading that demands an iron-willed conductor who is a great technician and an orchestra of exceptional precision. One has but to listen to the propulsive energies of the boisterous finale to appreciate the quality of musicians necessary for such brilliant playing.

The sound here is of huge, almost overwhelming proportions. The acoustic perspective is fairly broad, miking moderately close-up, the whole adding up to that happy blend of orchestral definition with presence that is so difficult to produce. Left/right directionality and center phantom channel combined to present an unbroken sonic panorama of the orchestra. Balance between the various choirs of the orchestra was superb. Dynamic range was quite wide, excellent transient response. All was exceptionally clean and a final bonus was the low noise levels of this splendidly processed EX-Plus tape.

Sibelius: Finlandia, Valse Triste, Swan of Tuonela, Tapiola. Herbert von Karajan cond. the Berlin Philharmonic Orch.
Ampex/DGG DGC9016, **4 tr., open reel, 7½ ips** ($7.95)

Herbert von Karajan has always been quite successful with the music of Sibelius. In this recording he gives us some outstanding performance of these popular works. The Valse Triste and the Swan of Tuonela are studies in virtuoso playing, and the English Horn solo in Tuonela is simply beautiful.

Karajan pulls out all stops in his rousing performance of Finlandia. It has propulsion, nobility and, above all, it is *exciting*. Tapiola receives one of the best performances ever, very properly atmospheric with absolutely stunning string work by the Berliners.

The sound is superb, and appropriate to each work. The two quieter works are clean and richly sonorous. Finlandia and Tapiola have sound of great power and presense. Some really weighty brazen brass here, and strings are almost searing in their intensity. The acoustics were spacious and miking nicely handled for excellent depth effects. Left/right directionality and orchestral balances in general were good. Dynamic range was exceptionally wide. Played at a fairly high level, the hiss was low, almost no print-through or crosstalk were noted. If you like Sibelius, you can't go wrong with this recording.

"Marching Band"

Sousa Marches: Band of the Grenadier Guards cond. by Major Rodney Bashford. London/Ampex LPL74103, **open-reel, 4 tr.** 7½ ips ($7.95)

This is a fine recording of its type. Although you might feel that if you've heard them all, this doesn't apply to band buffs, whose numbers are legion. They make the fine distinction of a marching band versus a concert band performing these works, maintaining that there are many differences. I feel that at least as far as recordings are concerned, it is mainly in matters of sound quality that affords a basis for comparison. The Grenadier Guards are without question a "marching band" and they perform these familiar marches with great verve and commendable precision. The program is typical with such favorites as "El Capitan," "Semper Fidelis," "Washington Post," "The Thunderer" and, of course, the inevitable "Stars and Stripes Forever."

The sound is excellent although lighter in weight than the kind afforded Fred Fennell in his Mercury recordings. Acoustics were not too spacious, but the miking was nicely handled and gave the music good presence. Good directional qualities with adequate center fill. Ensemble well-balanced, projection good, except for the bass drum which was a bit lacking in weight and prominence. Other percussion quite clean, brass and woodwinds nice and bright. At moderately high-level playback, hiss and crosstalk were low, some print-through noticeable.

Ormandy/Ives

Ives: Symphony No. 1 in D Minor; Three Places in New England. Eugene Ormandy and the Phila. Orch.
Columbia MQ991, open reel, 4 tr., 7½ ips ($9.95)

One can dispute the merits of the various conductors who have espoused the cause of Charles Ives, in an endless academic exercise. Ormandy has his faults, but I like what he does with this music, especially his "Three Places" reading. Perhaps I'm influenced by the splendid sonorities of the orchestra. At any rate, I'll be hard to convince that there is a better sounding version.

The Lighter Side

Swing Is King: Ted Heath and His Music.
Ampex/London LPL74104, open reel, 4 tr., 7½ ips ($7.95)

This isn't for those under thirty, but if any of that breed would like a sampling of what made their "square" parents "blow their cool," this tape will serve admirably. They are all here . . . those numbers that will always be equated with the thing called "swing," and so closely identified with certain "big bands." "Flying Home," "One o'Clock Jump," "Woodchoppers' Ball," "Sing, Sing, Sing" and eight other nostalgic flights back to the late '30's and '40's make up the program played with

appropriate swingin' style by Ted Heath . . . himself a refugee from that era. The sound is typical Phase Four, very live and BIG, with much exaggeration of directivity and reverb, but nonetheless quite clean and of considerable impact. Pleasingly low hiss, but some audible print-through and crosstalk.

Joel Grey: "George M!"
Columbia OQ1023, open reel, 4 tr., 7½ ips ($7.95)

This show received rave reviews, and from the evidence here they were well deserved. It has plenty of drive and zip, Joel Grey is perfectly cast as the brash George M. Cohan and the supporting cast would be hard to fault. The Cohan songs are a virtual roll call of Broadway. In spite of all these virtues, after awhile I found the songs repetitious and the whole thing rather dated. Heresy? Maybe so, or maybe I'm not as old as is necessary to fully enjoy this music. Good sound throughout, although I thought the voices somewhat peaky. Excellent stage movement. Some print-through and crosstalk, but the hiss level was very low . . . in fact this is one of the quietest tapes I have encountered.

Walter Wanderley: Kee-Ka-Roo.
Ampex/Verve VVC8739, open reel, 4 tr., 7½ ips ($7.95)

I don't know whether Kee-Ka-Roo is the call of some jungle bird, or whatever. There is some sort of weird sound that ties in perfectly with Walter Wanderley's exotic latin, yet jazzy rhythms. As usual this group plays with its typical bouncy ebullient drive and almost nervous propulsion. The sound is very bright, recorded quite close-up, but with just the right touch of reverb that combines for excellent presence. Little hiss, slight amount of print-through, crosstalk no problem.

The Enoch Light Singers.
Ampex/Project 3 PJX 5021, open reel. 4 tr., 3¾ ips ($5.95)

The Enoch Light Singers are not identified on this tape . . . which is a shame, for they certainly deserve credit for some excellent performances. They sing with considerable expression, with precise diction and are always articulate . . . a refreshing change from some of the dismal mumbling that passes for choral work these days. As the title indicates (and with justification for once) this fine group sings such "smash" hits as "It Must Be Him," "Love Is Blue," "I Say a Little Prayer," and "Up, Up, and Away." Eight other reasonably current hits round out the program, including a particularly attractive version of the "Ode to Billy Joe." Good vocal/orchestral balance, fine ensemble work. The reverb was moderately applied and the miking fairly close, adding up to good definition and presence. Hiss was moderate at fairly high-level playback. There was occasional print-through, almost no crosstalk. Nothing earth-shaking here, but a very pleasant tape, well above the average of its type.

Hugo Montenegro: Music from "The Good, the Bad and the Ugly," "A Fistful of Dollars," "For a Few Dollars More" · Music to Spy By.
RCA Victor TP3-5052, open reel, 4 tr., 3¾ ips ($9.95)

"The Good, the Bad and the Ugly," "A Fistful of Dollars," and "For a Few Dollars More," is a trilogy of films noted for their excessive and graphically displayed violence and gore. They were also characterized by music composed by Ennio Morricone of Italy, that caught the public's fancy and produced several hits. The music is rather sensationalistic and filled with tricky effects, that afford a field day for the sound engineers. A super multi-mike mix here with much spotlighting of instruments and deliberately overblown reverb. A lot of it is pure corn, but one cannot but admire the masterful recording job. Much the same can be said of "Music to Spy By," a compilation of TV and movie themes reflecting the current emphasis on spies and super-heroes. Thus we have music from "I Spy," "The F.B.I." and "Get Smart" along with "Thunderball," "Goldfinger" and the "James Bond Theme," and others. The engineers pull out all stops here and the result is almost like a short course in modern studio recording. Even with the restrictions of the 3¾-ips speed the sound is ultra-brilliant, with razor sharp definition and overwhelming presence. Like olives, one must develop a taste for this sort of thing . . . for those who are *muy simpatico*, this tape should make them flip!

OTHER RECOMMENDED TITLES

Sergio Mendes/Favorite Things.
Ampex Cassette ALX58177 ($5.95)

Tony Mottola/Latin Love-In.
Ampex Cassette PJX55010 ($5.95)

Roger Williams/More Than a Miracle.
Ampex Cassette KTX53550 ($5.95)

Something New/The Glenn Miller Orch.
Epic 8 Track, N1810116 ($6.95)

Bobby Hackett/A String of Pearls.
Epic 8 Track, N1810114 ($6.95)

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I certify that the statements made by me above are correct and complete.

CHARLES G. MCPROUD, Publisher

LETTERS

(Continued from page 16)

Electronic Organs

• I like your article on electronic organs in the September issue of AUDIO.

Perhaps there's something I don't understand about the various methods of tone generation, but isn't Electro-Voice's system of engraving wave forms on a stator plate a different method that you didn't discuss? That's the invention of Dr. Jean Dereux. Or is that method similar to the Compton's that you mentioned?

JOHN R. EMBLEN
Plainfield, Ind.

Author's Reply:

• The Electro-Voice company has been active in two distinct areas of the electronic organ market. When I prepared my original books on the subject (Sams' *ABCs of Electronic Organs*), the company had only recently introduced the home models which featured the chord coupling I described there, using the lower keys of the manual, with electronic oscillators and dividers for tone generators.

These organs were discontinued a couple of years ago and, knowing this, I did not check with Electro-Voice when I started to prepare the present series. At that time I was unaware that the company had more recently entered the larger organ market, using the type of generator to which you refer for organs usually installed in churches or public buildings. However, I understand that they are now discontinuing this production also.

To answer the technical aspect of your query, these larger Electro-Voice organs used a scanner that worked very similarly to the Compton. The difference was in the technique used for producing the circular tracks, which they did under license from the French inventor you named.

NORMAN H. CROWHURST
Gold Beach, Oregon

Capacitor PC Mounting

• Re your answer in February 1968 *Audioclinic*, the 4500 μF, 50 V capacitors are most probably CG452U50D1's, which are 2″ diameter x 4⅛″ high. This is a little big for printed-circuit board mounting by the terminal posts. The main point is, however, that normal practice is to use "mounting rings," which are a standard Mallory part, cost 18¢ for this size, and provide a wide, three-point mount. Sprague has them also.

WALT FERRIS
Palo Alto, Calif.

ELECTRONIC ORGANS

(Continued from page 80)

Vibrato

A later installment will go into details about vibrato, but applying a 'wobble' to the notes also relates to the problem of frequency stability, which we will consider here. If all the tones aren't precise, wobbling them a little, which vibrato does, will disguise the error. Some of the cheaper organs rely on this, to "get away" with poor tuning stability.

But even if your tuning is perfect, you'll want vibrato. An electronic organ doesn't sound right without vibrato, for most of the music you'll play. Some church or cathedral type music may sound right without vibrato, but you'll probably want something more versatile than that. This brings in the question of how you'll get your vibrato. The idea of vibrato by phase shift, introduced in the previous installment, is very attractive, in theory, and quite possible. But it isn't easy, except at the lower frequencies.

Suppose you phase-shift a lower frequency, say, 55 Hz. Its period is 18 milliseconds. If this is advanced a quarter of a period during half a second, the average period will be reduced and frequency momentarily increased.

Half a second of the original frequency would contain 27.5 periods. It will now contain 27.75 periods, or the frequency will swing up to 55.5 Hz average, with a peak of about 55.8 Hz. In the next half second the number of periods will be reduced, so the frequency drops to about 54.2 Hz. Thus the nominal 55 Hz will swing from 54.2 Hz to 55.8 Hz, or 3.2%, total of about half a semitone (quarter of a semitone deviation). This is quite an audible vibrato.

But now do the same thing with, say, 880 Hz, which is not the top of the keyboard by any means. With a one-second vibrato rate again, a single phase-shift stage can shift only from 879.2 Hz to 880.8 Hz, which is ± 0.06%, a deviation only the very best ears can barely detect. You'd never know you had vibrato!

To use the phase-shift method, multiple stages are needed, so that total phase shift can run several cycles or periods. If we rule that out as too complicated for us to tackle, we are left with methods that change the tuning of the oscillator. Organ makers do this in one of two ways.

Since semiconductors came into vogue, a popular way is to change the tuning of the oscillators by changing the bias on a diode that, when conducting, shunts an extra capacitor across the tuned circuit (Fig. 3-6). This is from a Lowrey circuit. When we discuss this, we'll find that care is needed in how the bias is changed, to avoid a jerky vibrato.

The other method in common use changes the bias of the oscillator itself. This is used in Fig. 3-5, which is typical of the method used in tube-type circuits. The actual mode of operation varies. The Conn, shown in Fig. 3-5, uses the oscillator almost in Class C. An oscillator working this hard, so the active element is inactive for the major part of the period, can best be considered as tuned circuit again.

Changing bias, which the vibrato voltage does, thus varies the amplitude more than frequency, although they are never completely separate.

Although the vibrato connection in Fig. 3-4 looks similar to that in Fig. 3-5, it works a little differently. The vibrato changes the bias in the base circuit, but it works through the change in collector voltage this causes. It combines frequency and amplitude shift.

Any vibrato circuit that shifts frequency can leave it permanently shifted, or out of time, if switching the vibrato off leaves an asymmetrical bias from the mid-point of the vibrato fluctuation. So the vibrato circuit must be designed to care for this.

This installment has covered most aspects of tuning (as regards stability, not the job of tuning the organ). Some modern organs use variable resistance for tuning, but the majority use a variable inductance. In the next installment, we'll move on to different ways of putting the possible electronic pieces together to make a whole organ.

CONTINUED NEXT MONTH

BEHIND THE SCENES

(Continued from page 12)

were great chambers, but they were costing us too much in rental so we put in EMT units. We found an interesting thing with the EMT units quite by accident. We put them right in the ballroom and found that we got acoustic coupling of the EMT's from the orchestra in the room, aside from the electrical drive, and it makes an unusual sound. I have an idea that ultimately, if we ever get time, I want to suspend plates in the studio over the various orchestral choirs . . . probably they will be much lighter in weight than the plate in the EMT, but likely of hardened steel. With them we'll pick up reverb selectively from the sound waves of the orchestra impinging directly on the driving unit. It may be impractical, but I'd like to try it.

How many inputs on your consoles? Does it vary?

FINE: Studio A has 14 inputs, Studio B has twelve. The new console for Studio A will have 24 inputs with a 16-channel output, plus a *semi*-computerized matrix mixing system, where you mix the monitor. In these multi-track systems, where you record wide open to the 8 or 12 tracks, where there is no mixing, they matrix the monitor, listen to every channel and mix in on the monitor so they hear some semblance of what they are doing, even though it has nothing to do with what is going on the tape. You also need an earphone mixing system, which we have in Studio A, because you are selecting from so many tracks a mixture of something to feed to the musicians to sel-sync against. Speaking of earphones, we are thinking of adapting an old Hollywood movie technique that is quite interesting. I should note at this point that all the 8- and 12- and 24-track recording of today is merely a modern upgrading of what are essentially old movie techniques.

You mean they actually did multi-track recording in those days?

FINE: I'll give you an example. In the old musical pictures of the Astaire/Rogers era, the orchestra recording was made, then the singing recording . . . all on optical push-pull tracks, and on synchronized equipment. This was played back to the artist who was photographed lip-synching to the sound. Loren Ryder of Paramount Pictures made an ingenious invention in the late thirties. He felt there was something missing about those people dancing in the musicals . . . the real sound of their clothing, the real sound of their dancing in the room that makes it lifelike, as compared to the lack of perspective in the picture where the sound was always at the same level, whether the dancers were close-up or far away. So he put a 150-kilocycle radio transmitter with a huge loop over the shooting stage and with a big antenna radiating from the roof of the building . . . remember, this was long before solid state, a typical brute force system . . . and made a device that fitted behind the performers' ears and couldn't be seen, with a germanium rectifier and a bone-conduction earphone . . . and just pumped enough "juice" into this system to rectify enough to drive the bone-conduction unit. Then they could play back the various orchestra and singing tracks to the performers, who could dance, while leaving mikes open on the stage which could pick up the real sound of the dancing and moving around, since the mikes naturally could not pick up the voice and orchestral tracks.

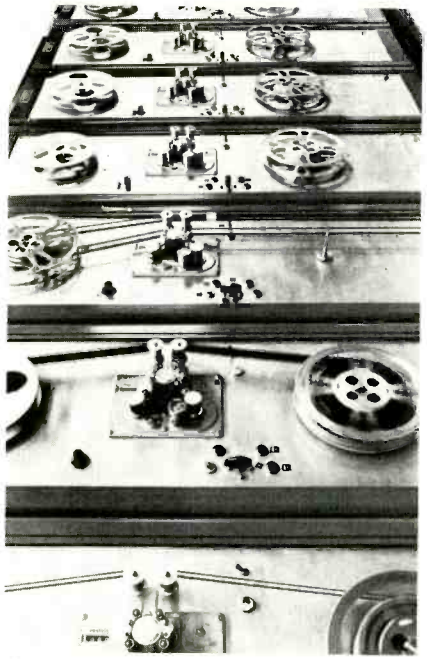

Fig. 3—Cutting Engineer George Piros is shown at the controls of a Scully lathe used by Fine Recording.

Fig. 4—35 mm magnetic film dubbing in Studio "C."

How do you intend to use this technique?

FINE: Well, we have such a maze of wires in the studio when we put out as many as 40 earphones on one of these 8-track pop recordings, that we are thinking of installing a similar radio transmitter earphone system. The only thing to figure out is how to keep the earphones from vanishing after each session.

I suppose you try to use some inexpensive phones, but there is a limit as to how much you can restrict quality.

FINE: Right, but even the little phones such as supplied with transistor radios cost 50 cents apiece. Nevertheless, we are going ahead with the radio earphone system for the flexibility it will afford.

Bob, in which of your studios do you prefer to do a rock-and-roll recording?

FINE: We are presently building Studio E, which among other things will have an especially dead room for this type of multi-track recording. However, as I have pointed out, due to the use of click tracks and sel-sync, even the ballroom is suitable.

On these rock-and-roll dates, the musicians are equipped with electric guitars and other electric instruments, each with its own amplifiers and speakers. I assume you don't pick up the sound of their speakers via microphone, but go into your preamps directly from their bridging outputs?

FINE: No, we don't go in direct. But it is wise to eliminate mikes. I've found that the only place you can get that raw sound peculiar to these instruments is across the voice coil of their speakers. You don't get the same thing out of their bridging amp output because it is before the power stage. So we are putting in a system that clips right to the voice coil. This picks up all the "distortion" that gives these instruments their "character" and would appear to be a product of the clipping of the output transistors.

Disc cutting has always been one of your major interests. What constitutes your present system?

FINE: Our stereo disc mastering equipment is of our own design. You'll be interested to know it was installed 9 years ago. We took a different philosophy about what is necessary to cut a stereo disc than generally prevails in the industry. For one thing, we use

200-watt-per-channel modified McIntosh's as our cutting amplifiers. And we use only the original Westrex 3A cutter.

Why is that, Bob? I've heard comment that "Fine ought to get modern and use the Westrex 3D."

FINE: We modify the suspension of the 3A ourselves. We don't use feedback. We use something else because we don't believe in feedback, especially the Westrex feedback system, because that is not a true way to correct mechanical motion.

That surprises me. Weren't you always an advocate of feedback in the days when you used the Miller cutter?

FINE: It is not the same feedback . . . the Miller used a *mechanical* feedback. There were certain rods in the cutter that had to be tuned. At any rate, with our present system we can put pulse information and high frequencies on a record without resorting to Conax equipment. [the Conax equipment is a frequency-selective limiting system] Almost every studio in New York uses Conax equipment because the Westrex 3D systems will not pass high-frequency, high-power information, so they use frequency compression systems to get the level on their records. Six or seven months ago we got a Westrex 3D cutter and set it up on another Scully lathe. We were asking ourselves whether we were behind or ahead in cutting technology and thought it was time for a general review of disc cutting. We set up the 3D without a Conax and fed it and our 3A system the same program material. As far as I was concerned, the 3D couldn't put what I wanted on the disc. The 3D exhibited that d.c. blocking which comes from feeding instability in the system . . . it just sounded nothing like we were doing. You couldn't put anywhere near the peak levels on the record that we could with our system.

I was under the impression that most companies using the 3D are using the HAECO amplifiers, which are reported to be quite good.

FINE: Yes they are, and the HAECO is quite an improvement over the Westrex amplifier, which is really rather primitive. But this still does not overcome the problems of the 3D. The 3D, by the way, was made with a looser suspension to reduce the power requirements. I don't believe in that. I feel that a cutter should be absolutely rigid. The Miller cutter had no movement of the armature at all . . . you actually distorted the metal with a magnetic current.

I notice you still use the Scully lathes. Do you anticipate any changes, perhaps to the new Neumann computer lathe, for example?

FINE: I think they are fine machines and are good for studios with a great amount of disc cutting, but they are concessions to individual mastering. In other words we took off all the Scully sensing equipment on our automatic variable pitch lathes. This would have made life easier having it, but it can't "think" how much the time factor is on the record or how loud you want the record to be. So you are making a concession whether you master with a Scully or Neumann or Ortofon automatic system. You never really make a record on this kind of equipment that has that little extra something in it that makes it a better product. I believe that when you go to the trouble and expense of making a symphony recording or any high-quality recording, it deserves to be cut onto a disc with individual attention to levels and dynamics and every other significant parameter. However, just so I won't be accused of sour grapes . . . if I had the amount of disc cutting that RCA and Columbia do, I would own an automatic Neumann too!

Bob Fine has often been accused of putting too much level on records.

FINE: It's true . . . I feel that the records should have a lot of level on them. I must confess we do get up to extreme velocities.

Up to 30 centimeters per second, I understand.

FINE: Yes, peaks of 30 cm, but in defense I challenge you to see a Teldec, a Grampian or a Westrex 3D cutter put that down on a record, and put it down with complex pulse.

Can present playback equipment cope with these high velocities?

FINE: Most certainly the better stereophonic arms and cartridges can, and even some lesser consumer equipment, which is getting better all the time.

In addition to all your studios you have a tape-duplication facility in the basement of the hotel. Is it the usual master playback with reel-to-reel slave machines, or do you use a common mandrel system?

FINE: We have a master and ten slaves in a special modification which is saving us a lot of money. Heretofore, if we were not running the kind of tape format the slaves were set up for, the machines were idle and non-productive. Now these ten slaves have a programmed head system. They have a 52-prong machined base, with a 52-prong plug-in head assembly which is changed for each tape configuration. On quarter-inch tape we have interchangeable heads for monophonic, two-track, and four-track. On cassette 150-mil tape, we have two- or four-track heads. The heads pick up the guidance of the type of tape and there is room to program the tensions on the machines and what kind of a master is being used . . . either loop or reel to reel. On the top of each head is a level and bias control . . . no more diddling with the bias controls on rear panels. So whatever kind of tape format we are asked to duplicate, it is a simple matter of plugging in the appropriate head, and this keeps the "down-time" on the machines to a minimum. We do cassette duping at 16 times speed ratio, reel to reel at 8 times speed, and 8-track cartridge is also duped at 8 times the speed ratio. The 8-track is made on another bank of ten slaves which we are going to convert to the plug-in programmed head system.

High hiss levels seem to be the particular curse of present cassettes. Can't anything be done to bring the hiss down to "tolerable" level?

FINE: There is a very critical bias-frequency and bias-level problem when running cassette dupes at 16 times the speed. There are also problems of stability with the tape and tape tensions on the duplicators. We are making progress in overcoming these problems. We are now looking into different types of bias systems. We are looking into different bias arrangements than just the bulk pumping of bias into a head to record. There has been a lot of saturated gap recording. I have no experience with it, but question the merits of this approach. I believe there will be technological advances that will eliminate the basic practical problems of bias on tape. Most present oxides are a holdback. In our duplication set-up we have tried many different kinds of tape, and only one tape works well at high speeds.

What are your feelings about the new DuPont chromium dioxide tape?

FINE: I think there is great potential in the Crolyn tape.

What about the reported abrasiveness of the tape to the head and the requirement for almost double the bias?

FINE: The bias problem exists because of the present configuration of the circuitry. We certainly can build new equipment to handle this, and I believe the abrasiveness can be licked.

Do you think tape and disc can continue

to co-exist in the music market?

FINE: Very definitely. I think both mediums have yet untapped potential. I am very optimistic about tape as a recording medium. With its constant lineal speed there is no diameter problem, as in the case of discs. Since most music has a loud finale, and this usually appears at the inner diameter of a record, we run into tracking difficulties. This has no relevance in tapes. Of course, one of the big problems in tape duplication is the number of generations we must go through before we get to the consumer product. What we get from a record company is usually a third generation working part. This is, in essence, library material which we can't cut up, so from that we make an intermaster, which is then compressed in dynamic range to give us a better signal-to-noise ratio with all the dubbing going on. Then the tape goes through program preparation, according to whether it is cassette, 8-track, etc., and a running master is made. This is your dubbing master for high-speed duplication. The result of all this is that we may have gone through six or seven generations. Which, incidentally, is what makes the Dolby system so useful in this respect. Naturally all this dubbing of tape cannot be compared with a record, which, at worst, is a third-generation product. Consequently, a good record is quieter than most pre-recorded tapes, and why the record still is so strongly competitive.

The cassette seems to have caught the fancy of the public. Would you venture some prognostications about the future of cassettes?

FINE: For one thing, I have no doubt that chromium dioxide tape is a superior medium for cassettes. I foresee that a whole LP record can be contained in a package half the size of the present cassette, running at a speed of a half-an-inch per second, that will give you a frequency response from 15 Hz to 20 kHz. There should be no problem with heads. The cassette will be "bubble" packaged, which should cost less than a cent and a half, and the total cost will be less than a record pressing. I envision this within 5 years.

What about signal-to-noise ratios?

FINE: It will be far superior to what we have now. You see, head technology is moving ahead at a tremendous pace. This is due to the stimulus of sophisticated telemetry systems in the space program, where they are doing incredible things with frequency response at slow lineal speeds. I am certain many of these advances will "filter" down to the consumer level. Æ

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