the authoritative magazine about high fidelity

# AUDIO

CDC

JUNE 1968 60¢



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*Also: A Custom Stereo Installation*

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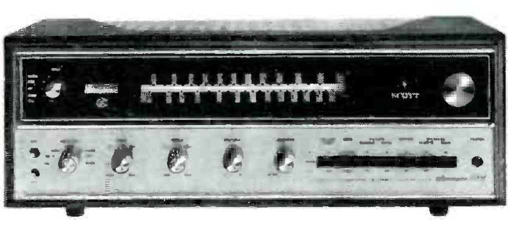



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June 1968 Vol. 52, No. 6

# AUDIO

Successor To RADIO, Est. 1917

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# Coming in July

**Audio/FM Test Instruments**—A roundup of test instruments used to examine audio, FM and stereo FM equipment.

**Electronic Crossover Networks Revisited**—C. G. McProud discusses how to modify his original three-channel electronic crossover construction project (Audio, February 1968) to create a two-channel unit. In addition, he provides information on how to change values of parts to obtain different crossover frequencies.

**Exotic Loudspeakers**—Al Fanning surveys loudspeakers other than moving-coil types that are used by some hi-fiers.

**Audio Noise Reduction**—Ray Dolby concludes his article on audio noise reduction by examining more aspects of the Dolby professional system and by discussing some practical applications.

**EQUIPMENT PROFILES:**

**Fisher** Model 200-T Stereo FM Receiver

**KLH** Model Twelve Speaker System

**Plus:** Regular monthly departments, music and record reviews, and more.

**ABOUT THE COVER:** The attractive cabinet shown on the front cover, designed by Jerry Joseph of Toujay Designs, Inc. for Edward Greene of Baldwin, L. I., N. Y., is approximately eight ft. long by eight ft. high. It was delivered in five sections, and said to take less than 30 minutes to be set up permanently. The equipment it houses includes: a Tandberg tape recorder, AR manual turntable, Dyna preamplifier, power amplifier and tuner, and a pair of Bozak B302A speakers. Also accommodated are 400 LP records and 300 reels of tape. Note that equipment is at comfortable operating height.

# 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.

## Silicon rectifiers

*Q. I am one of the nation's first lifetime subscribers to* AUDIO. *I think my membership card is No. 89. I have read and enjoyed your column for many years.*

*I am currently interested in building oscilloscopes. I have had some success, but I run into trouble with the high-voltage supplies. I obtain the high voltage by combining the secondaries of a couple of power transformers in series. The problem is that in the circuitry, this high voltage is applied to the filament windings. These windings are not insulated for this high voltage. The result is, after a short time, a breakdown.*

*I am wondering about the possibility of using silicon rectifiers in series instead of vacuum tubes as rectifiers, and thus eliminating the need of the filament voltage. I get the impression from my reading that it is possible to handle higher voltages than individual silicon rectifiers can withstand by connecting them in series. I have been strictly a tube man so far. The only time I tried to use the silicon rectifiers, they overheated; it just did not work. Why, I do not know.*

*For instance, I have some silicon rectifiers rated at 750, 500 mA. Could I connect three of these somehow to handle a negative 2,000 volts?*

*Are there any disadvantages in using these silicon rectifiers? They are so small and inexpensive and do not require filament voltage. It would seem that no one would want to bother using tube rectifiers.*—Rev. Francis J. Jann, Belfast, N. Y.

A. From your letter I gather that the insulation between windings of one or the other of your combined power transformers is breaking down. What you can do is simply employ a separate filament transformer to heat the rectifier tubes. Be sure, however, that this transformer has sufficient insulation between its windings and its core to avoid breakdown at the high voltage used.

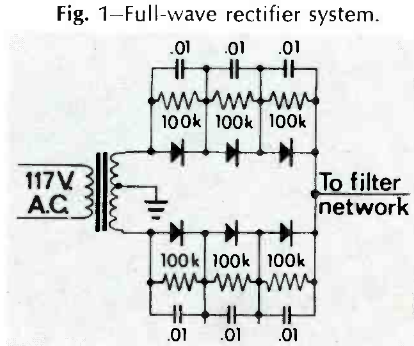
You can, of course, employ silicon diodes. This is the better, easier course to follow. When series-connecting them, though, you should take the same kind of precautions as you would if you connected capacitors in series to obtain a higher voltage breakdown rating. You should shunt high-value resistors across each diode. This will equalize leakage currents and prevent one diode in the string from taking most of the burden. Further, each diode should be equipped with a shunt capacitor of perhaps 0.01 $\mu$F. This serves to hold down certain transient peaks which come along and often cause diodes to fail. A full-wave rectifier system employing these techniques is shown in Fig. 1.

From your letter it is not clear what circuit you are using: bridge, voltage doubler, etc. Therefore, I can only give you some general guidelines on how to make your circuit work in the best possible manner.

If your circuit employs a full-wave rectifier from which you obtain 2,000 volts, you will need at least six diodes in series in each leg of the circuit. I am, of course, assuming you will be using the diodes of which you wrote in your letter, those having a 750-volt PIV rating. Actually, these six diodes will handle a peak voltage of 4,500 volts, dangerously close to the voltage which may be present in this circuit during times that the diodes do not conduct. Further, one or more diodes in the string may be incapable of handling their rated peak voltage. Therefore, I suggest that you use eight diodes in series on each leg of the full-wave bridge.

I suspect, though, that you are probably deriving your high voltage by means of a positive and negative power

**Fig.** 1—Full-wave rectifier system.




supply system, with chassis ground serving as the midpoint of the two supplies. Because there is little current drawn from this high voltage supply, I would assume that you might be using a half-wave rectifier system for each power supply. The typical halfwave rectifier system is shown in Fig. 2. This circuit shows a supply which will provide a voltage which is positive with respect to ground. You would have to reverse the connections to each diode, and also reverse the polarity of your electrolytic filter capacitors if this same circuit were to be used to give you the


counterweight
Sliding counterweight is infinitely adjustable, makes precise dynamic balancing practical; locks into place, isolated in rubber from the arm
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Patented anti-skating control, of spring-free sliding weight design, has 1/2 gram calibration to nullify side pressure on stylus and groove walls.
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Insert the most critical, most sensitive cartridge in the Garrard SL 95, and be assured it will travel the intricate convolutions of the grooves easily and flawlessly. The advanced, ultra-low mass, gyroscopically gimballed tonearm system has just three controls, to balance it exactly to the weight of the cartridge, adjust it precisely to the recommended stylus force and counteract the natural skating tendency, providing perfect tracking, distortion-free reproduction.
The SL 95 gives your records a perfect ride, too. The synchronous Synchro-Lab Motor™ matched kinetically to the oversized, balanced turntable, guarantees absolutely constant speed, unwavering pitch, freedom from rumble. Simplified cueing and pause control, and the exclusive safety record platform, protect your records both in manual and automatic play, making the SL 95 Automatic Transcription Turntable the ultimate in performance and dependability. Price: $129.50, less base and cartridge. Other Garrard models as low as $37.50.
For a complimentary Comparator Guide to all models, write Garrard, Dept. AC-1, Westbury, N.Y. 11590.
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World's Finest
stylus pressure
Top-reading stylus pressure gauge sets tracking force precisely with audible/visible click stops at 1/4 gram intervals.
cartridge
New cartridge clip guarantees safe, convenient mounting, in perfect alignment. Tone-arm and "shell" are of rigid, resonance-free, one piece construction.
TURNTABLE
England
95
STOP
START

negative portion of the supply voltage.

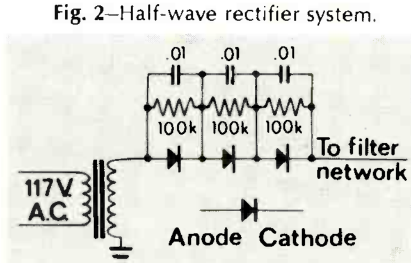
The complete power supply circuit, employing both a negative and a positive full-wave supply, is shown in Fig. 3.

Take into account that while the voltage under load may be 2,000 volts, the voltage without load (such as would be true during times of tube warm-up, etc.) might be close to 3,000 volts. Hence, the filter capacitors must be rated to handle at least a peak voltage of 3,000 volts (and it is best to provide a safety factor). Therefore, your filter capacitors should be rated at 4,000 volts PIV.

I do realize that the oscilloscope tube will not provide much of a load on your power supply. It may be, therefore, that you took this matter into account. Possibly your power supply will deliver 1,500 volts under load, but will provide the necessary 2,000 volts or thereabouts when only lightly loaded by the CRT.

The symbols typically employed to denote the anode and cathode connections of a silicon diode are shown in Fig. 2. These are what you will find in schematic diagrams. How the particular diode you plan to use is marked is anybody's guess. Because of the lack of a common standard in this matter, measure each diode with an ohmmeter. When the meter is connected to read the lowest resistance, the positive meter lead is the plate, or anode, of the diode. The negative meter lead will be the cathode connection of the diode. I can't impress on you strongly enough to be sure you know which is the positive meter lead. I very nearly came to grief just the other day because I believed that the common meter lead of the VOM I was using was negative for all functions. It wasn't, and I had to reverse the connections to 5 diodes.

**Fig. 2**–Half-wave rectifier system.




If you know how to wire tube rectifiers, you can, most of the time at least, wire up the silicon jobs in the same way. Naturally, you must watch out for the peak voltage. If the peak voltage in a given circuit is more than can be withstood by the diodes, more than one diode can and should be connected in series, as I described earlier.

In addition to the advantages of no heater voltage being required, such diodes also offer the advantage that they have a lower voltage drop than do tube rectifiers. Thus, the power regulation of the power supply is better with these units than it is with tubes.

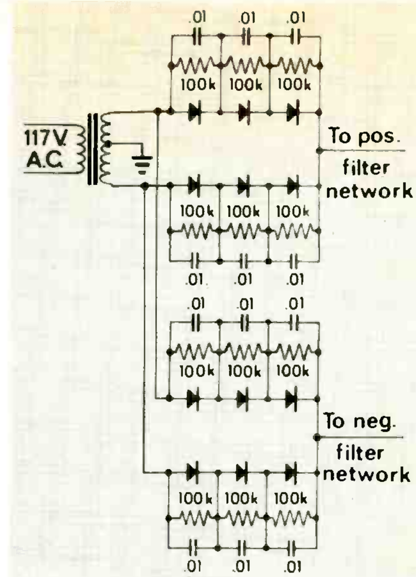



**Fig. 3** – Positive and negative full-wave rectifier system.

Like everything else, there are a few disadvantages. If you replace a tube rectifier with silicon diodes, the power supply's higher voltage may be more than certain circuit elements can withstand. Further, because silicon diodes have such a low voltage drop, they will, when connected to low-impedance sources (such as the 117-volt power line), draw very high current. Often the current drawn will be beyond the capability of the diode. These diodes, therefore, must be protected by the insertion of a series, current-limiting resistor. Even with this resistor included, the internal resistance of the power supply will be better than it would have been if tube rectifiers had been used. The value of this resistor for a.c. power line service is usually in the order of 30 ohms for equipment such as table radios.

A disadvantage which some people have encountered results from the diode's ability to conduct and then switch to a nonconducting stage extremely rapidly. While this characteristic makes diodes fine performers as switches, this situation can, and often does result in the production of transient voltage peaks which will damage the diodes and/or introduce annoying buzzing into certain pieces of audio equipment. The shunt capacitors shown in Figs. 1 to 3 are designed to eliminate these problems.

Though the story of silicon diodes is too long to tell here, it is my hope that this information will be sufficiently detailed to enable you to design more efficient, cooler-running power supplies. Æ

# When Stanton engineers get together, they draw the line.

The frequency response curve of the new Stanton 681 Calibration Standard is virtually a straight line from 10-20,000 Hz.

That's a guarantee.

In addition, channel separation must be 35 dB or greater at 1,000 Hz. Output must be 0.8 mv/cm/sec minimum.

If a 681 doesn't match these specifications when first tested, it's meticulously adjusted until it does.

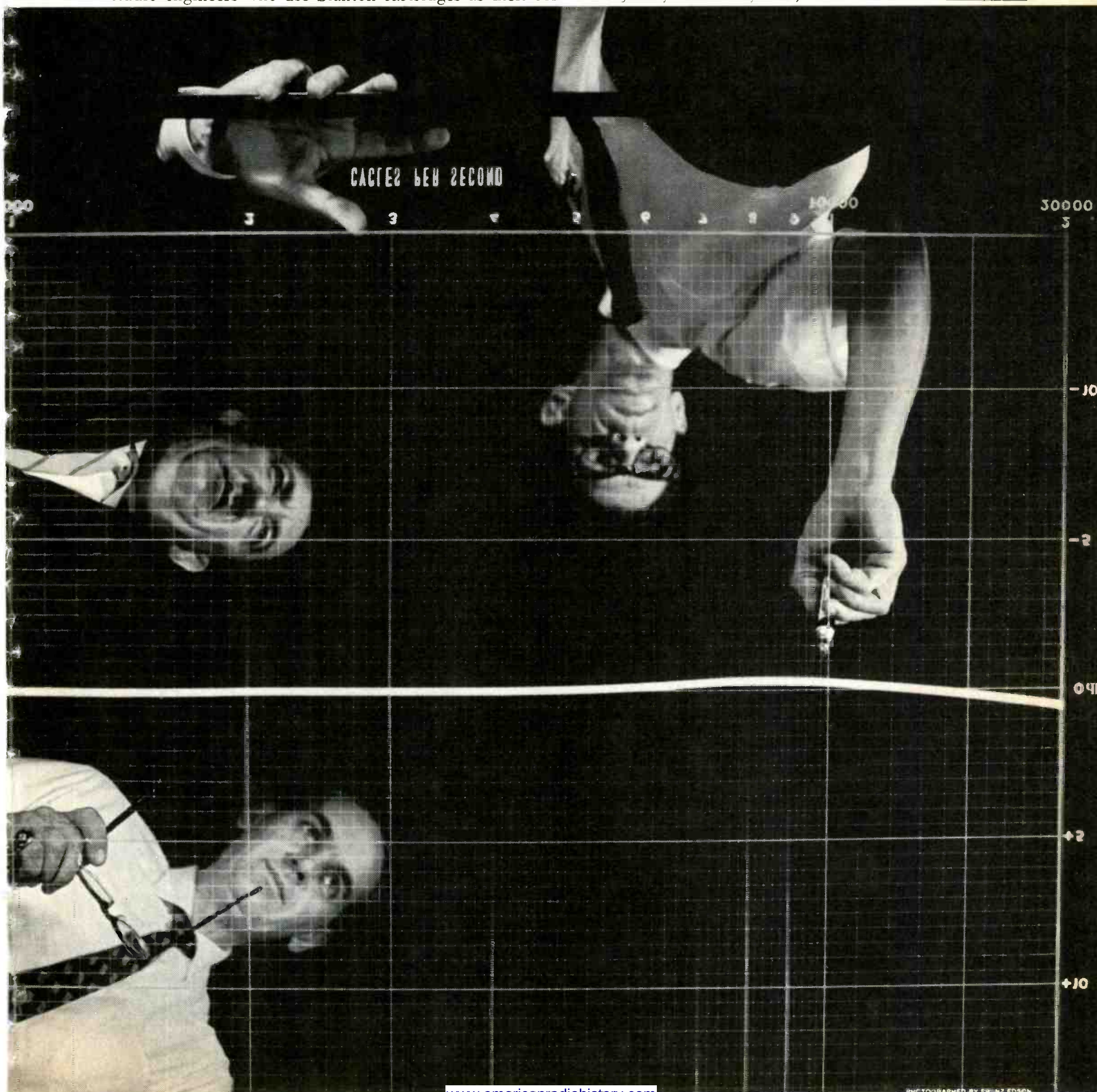
Each 681 includes hand-entered specifications that verify that your 681 matches the original laboratory standard in every respect.

Nothing less would meet the needs of the professional studio engineers who use Stanton cartridges as their reference to approve test pressings. They must hear exactly what has been cut into the grooves. No more. No less.

But you don't have to be a professional to hear the difference a Stanton 681 Calibration Standard will make, especially with the "Longhair" brush which provides the clean grooves so essential for clear reproduction. The improvement in performance is immediately audible, even to the unpracticed ear.

The 681 is completely new, from its slim-line configuration to the incredibly low-mass moving system. The 681A with conical stylus is $55.00, the 681EE with elliptical stylus, $60.00.

For free literature, write to Stanton Magnetics, Inc., Plainview, L. I., N. Y.

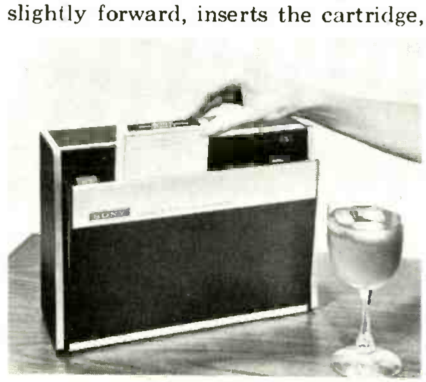


# What's New In Audio

## Eight-track cartridge recording

Here's a real innovation that will make many 8-track cartridge tape machine owners very happy—a device capable of recording on standard 8-track cartridges from a number of sources!

Introduced by Sony/Superscope, the unit, Model TC-8, has inputs for tape recorder, phono or FM multiplex. To record, the user tilts the cartridge panel slightly forward, inserts the cartridge,



and presses the "record" button. Power is automatically turned on when the cartridge is inserted. If the cartridge is inserted improperly, a cartridge-alignment indicator lights up and the TC-8 will not operate until it is properly reinserted. The unit features an automatic recording control, automatic shutoff, and an indicator light which identifies each channel being recorded. Price is $99.50.

Check No. 10 on Reader Service Card

## Pioneer 170-watt receiver

Pioneer's new SX-1500T AM-FM stereo receiver features 170 watts total music power (IHF) with a 4-ohm load; 140 watts total with an 8-ohm load (the usual speaker impedance). FM sensitivity is 1.7 microvolts, while other FM specifications are: capture ratio, 1 dB at 98 MHz; frequency response, 20 to 70 kHz ±1 dB; signal-to-noise ratio (IHF), 65 dB; harmonic distortion, under 0.1%; channel separation, 37 dB at 1 kHz.

The receiver, which includes an FET front end and integrated circuits in the i.f. section, has a full complement of controls and switches. These include high- and low-frequency filters, loudness contour control, tape monitor switch, and bass and treble controls for each channel.

Other important features include: two magnetic phono inputs, tape recording/playback jack, stereo headphone jack, automatic stereo indicator and a signal-strength tuning meter. Price is $345.00.

Check No. 12 on Reader Service Card

## AKG FET Condenser Microphone System

A new AKG modular condenser microphone system has been introduced by North American Philips Company, its distributor. Built around a new C-451E preamplifier, which incorporates field-effect transistors, it has interchangeable pickup capsules: CK-1, cardioid; CK-2, omnidirectional; CK-6, switchable from cardioid to omnidirectional to figure-eight; and CK-9, "shotgun" attachment.

In addition to being able to be powered directly from most amplifiers, a dual a.c. power supply for two microphones is available (Model N-46E), as well as a battery power supply for a single microphone. The latter employs a regular 9-volt battery.

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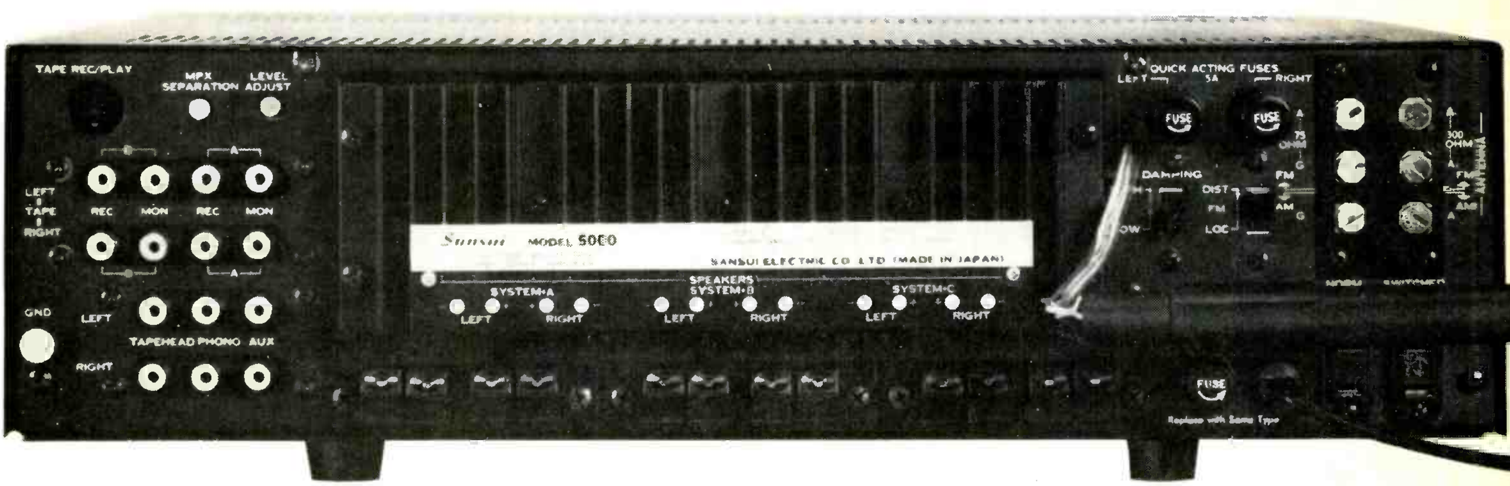
# Back Power!




Take a close look at the back of the powerful, exciting, Sansui AM/FM Stereo 5000. You'll see the inputs for 3 pairs of stereo speaker systems that can be played individually or in pairs—engineered quick holding plugs that eliminate the need for cumbersome clips; selective monitoring for 2 tape decks so that you can monitor while you record. Even the inputs for phono, tape, and aux. are grouped for easier access and to reduce the chance of wires accidentally touching. ☐ The Model 5000 Receiver features FET FM front end and 4 Integrated Circuits, with a set of specifications that exceed Sansui's unusually high standards — 180 watts (IHF) music power; 75 watts per channel continuous power; FM tuner sensitivity of 1.8$\mu$V (IHF); selectivity greater than 50 db at 95 MHz; stereo separation greater than 35 db; amplifier flat frequency response from 10 to 50,000 Hz. ☐ The front of the Sansui 5000? See it at your franchised Sansui dealer. Price **$449.95**

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Sansui Electric Company, Ltd., Tokyo, Japan • Electronic Distributors (Canada), British Columbia


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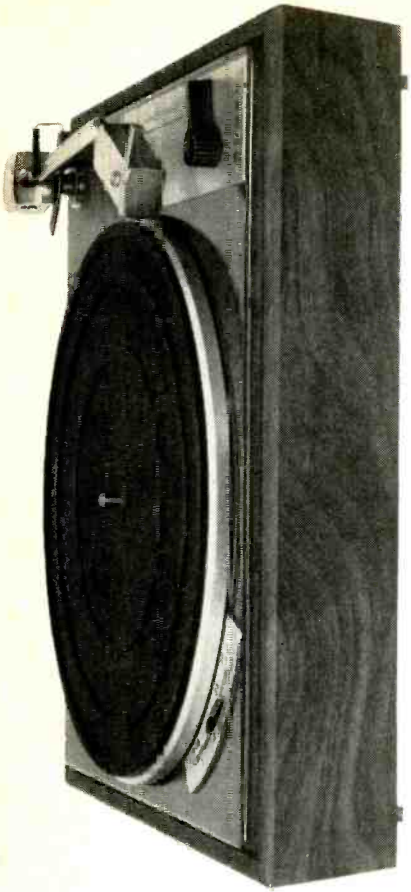




# Letters

• In the article "ABZ's of FM," April AUDIO, there is an error in Table I. This refers to the various classes of FM stations as defined by the FCC but it actually lists the classes of AM stations in the standard broadcast service.

Section 73.206 of the FCC Rules and Regulations defines commercial FM broadcast stations. Basically these are:

| Class | Max. Effective Radiated Power | Antenna Ht. Above Avg. Terrain |
|---|---|---|
| A | 3 kilowatts | 300 feet |
| B | 50 kilowatts | 500 feet |
| C | 100 kilowatts | 2,000 feet |

As author Feldman points out, the antenna height of an FM broadcast station has a lot to do with the station's range. To correct for this and put all stations on a competitive basis, as well as minimize allocation and interference problems, the FCC engineering standards provide for equivalent coverage. When the antenna height is above the standard specified for the class of station, the effective radiated power must be reduced according to a formula. On the other hand, no increase of power over the maximum is permitted if the antenna is below the standard height. Average terrain, by definition, is the average of all elevations between two and ten miles from the antenna. It is generally calculated along eight radials which must include the most prominent topographical features within the mileage limits.

Since very few antennas are at the same height, station powers have many different values. This has led to confusion on the parts of both listeners and advertisers because some stations publicize their actual licensed rating whereas others use the maximum ceiling value for that class of station. While these larger numbers may appear to make the station seem more powerful, the simple truth is that all FM stations operating with maximum facilities have equivalent coverage.

HAROLD A. DORSCHUG
Hartford, Conn.

• On page 34, Table I of "ABZ's of FM" purports to show the class, channel and power of FM stations. The table is actually an obsolete version of that information for AM stations.

Immediately below the table, the first sentence in the paragraph is incorrect. It reads: "Power will be determined by elevation of antenna, since the higher the elevation of antenna, the greater the coverage or range of transmission and reception." The sentence should read: "Received field intensity will be determined . . ." Height of the transmitting antenna has no effect on transmitter power or effective radiated power, but only on received field intensity. All other parameters being constant, the higher the antenna, the greater the field intensity at a given point.

On page 75, the first complete sentence reads: "As an example, a transmitter having a power of 20 kW will be received with a signal strength of 1000 microvolts per meter at a distance of 32 miles if its transmitting antenna is 500 feet above sea level." There are three misleading points in this sentence and in the rest of the paragraph.

The figure of 1000 $\mu$V/m at 32 miles has been computed from the chart in Fig. 1 of FCC Sec. 73.333. Power is defined therein as *effective radiated power*, not transmitter power. ERP is equal to transmitter output power, less transmission-line loss, times antenna gain.

For purposes of computing FM field intensity, antenna height is always specified in feet *above average terrain*, and never above sea level. The two are quite different, and computing average terrain around an FM transmitter site is a major task.

Finally, the author has omitted to mention the fact that Fig. 1 of FCC Sec. 73.333 shows received field intensity *only* when the receiving antenna is 30 feet above the ground. The antennas of many FM listeners are probably only around 5 feet above the ground.

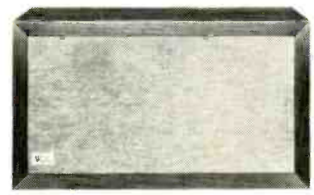
PAUL NORMAN
New York, N. Y.

*Here is the author's reply:*

The AM Station Power Classifications were incorrectly copied as Table I, as pointed out. With respect to comments on the relationship of power to elevation height, I did not want to imply that effective radiated power is in any way related to the height of the antenna. Mr. Dorschug read me correctly, as indicated in his third paragraph.

The comments about ERP and average terrain are correct, although my reference to sea level was made only in illustrating a *comparison* between two antenna heights, not to setting a new standard of average terrain measurement. In the May issue of AUDIO, in a discussion of receiving antennas, the point is made that elevation of the receiving antenna has a direct bearing on received field intensity of signal strength.

LEONARD FELDMAN



# AR-3a

## *Acoustic Research announces a new speaker system.*

In 1959, our first advertisement for the AR-3 stated, "it has the most musically natural sound that we were able to create in a speaker, without compromise." This judgment was supported by distinguished writers in both the musical and engineering fields. Hirsch-Houck Laboratories, for example, agreed that "the sounds produced by this speaker are probably more true to the original program than those of any other commercially manufactured speaker system we have heard." For nearly nine years the AR-3 has been the best speaker we could make.

However, technical development at Acoustic Research, as at many companies in the high fidelity industry, is a never-ending search for improvement. After much effort we have found a way to better the performance of the AR-3. The new speaker system, the AR-3a, has even less distortion, more uniform dispersion of sound and still greater power handling capability. The improvement can be heard readily by most listeners; it has been brought about by the use of newly designed mid-range and high-frequency units, and a new crossover network. Only the woofer and the cabinet of the AR-3 are retained in the new system. The AR-3a is priced from $225 to $250, depending on cabinet finish, and is covered by AR's standard five-year speaker guarantee.

Detailed information on conversion of an AR-3 to an AR-3a is available from

**ACOUSTIC RESEARCH, INC., 24 Thorndike St., Cambridge, Mass. 02141**

Check No. 9 on Reader Service Card

# AUDIO, ETC.

EDWARD TATNALL CANBY

## The Visible LP

NO MATTER WHERE you look these days, or where you listen, you'll find audio entangling itself in other nearby arts, crafts, and sciences. It's what's going on everywhere.

Take an item that has been buzzing around in my head ever since I saw and heard a demonstration last autumn (another product of the AES Convention). It is tentatively called *Phonovid* and was developed experimentally (so they say) by Westinghouse engineers.

The reason Phonovid interested me was that its visible product and, in fact, its whole central *raison d'être*, is an LP record. Not a special one, but a very ordinary, completely standard, commercially pressable LP. On the surface (and ignoring the label) you couldn't tell it from a couple of billion others. It comes out of a standard pressing plant, cut right along with all the others on standard cutting heads, off standard recorded tapes. It plays on the usual good-quality LP playing equipment, without any special differentiation—ordinary motors, cartridges, styli. All quite routine.

But what "plays" from this LP record, with a bit of extra circuitry hooked into the system, is *pictures*. Plus a sound track, in sync.

Put the record on an ordinary LP player with this system, lower your ordinary stylus into the wholly standard groove, and you get a long, measured succession of still pictures in black and white on the face of a TV set connected to the system. Up to 400 "slides," to use the nearest conventional terminology, projected neatly one after another from the LP record with time to look at each one and listen to the accompanying sound.

And, of course, you may hook in as many TV sets as you want, if you need to. That might be quite a number if, say, you were a school. Or a school system. Or a business with a host of audio-visual messages to get over to a great many people. Or a specialized training course, mass-production type. Or, maybe, the U. S. Army or something.

In other words, here is a complete built-in lecture on an LP record—the lecture itself, the audible message, plus really copious quantities of pictures, each one "frozen" on the screen long enough to look at, as the sound track does the describing. *Good* TV stills, too. I saw them on multiple rows of TV viewers down the sides of an auditorium. They seemed to me to be just as clean, just as steady and sharp and undistorted, as any normal black-and-white TV images you're likely to achieve on standard commercial sets, either broadcast or closed-circuit.

All this off a totally standard LP!

You see how clever the thinking was. The actual recording system (I'll get to it) is not really the vital point. The LP record itself came first, and determined the parameters. Plus the standard TV screen. Anyone who knows TV and cathode-ray technology can figure out how you might derive pictures from a disc groove. Nothing revolutionary at all. *Except*, in Phonovid, the specific choice of working parameters. What matters, you see, is *the medium*. (Aren't we supposed to understand, today, that "the medium *is* the message"?)

The LP record is one of the simplest, the cheapest existing vehicle for mass-scale information that we now have. And it's available.

And the same with the associated TV. Standard stuff. Though TV isn't exactly cheap nor inherently simple, it is also very definitely *available*. It exists. It is widespread and its presence is increasing everywhere, and most especially in the very places—schools, industrial training centers, sales meetings, etc.—where this Phonovid system might be most useful.

With standard equipment at either end, Phonovid is off to a flying start. All you need is the middle circuitry, the minimum that can be added to operate within the standard parameters at the two ends. That was the experimental project.

Ah, how about all those "limitations"? There are plenty, if you insist on looking at it that way. But in this sort of fun-and-games project you start at the far end and work backwards.

Phonovid is quite unlike, say, CBS's new EVR (Electronic Video Recording), which introduces its cannily designed new equipment to meet very different basic parameters—a maximum compactness of information content on a much more demanding scale, recorded TV in full motion *and* in color. Phonovid's LP disc contains no TV motion. No color. Just pictures. And sound.

As it has worked out, Phonovid projects slowly. Much too slowly for direct continuity. It gives you, as the LP record plays, a good TV picture every six seconds (though you may hook them together for longer shots, when desired). And so you see what we have. This is a system technically related to the close-spaced moon shots we all saw. Successive views, each a still picture, each advanced in "motion" over the previous one. But those were more rapidly projected. At a six-second spacing even that sort of semi-motion is out of the question.

Why stills only? To "move" a TV image you need a very wide bandwidth, as the videotape people know only too well. To get TV in motion onto tape you must resort to all sorts of inherently complex mechanisms providing in one way or another the high tape-to-head speed that is essential.

Your LP record has a hopelessly limited bandwidth from the *moving* TV viewpoint. But given a bit of time, you discover things aren't so bad. All you have to do is to scan your picture more slowly. Much more slowly. And store up the information as you go along, until you want to use it. A memory device.

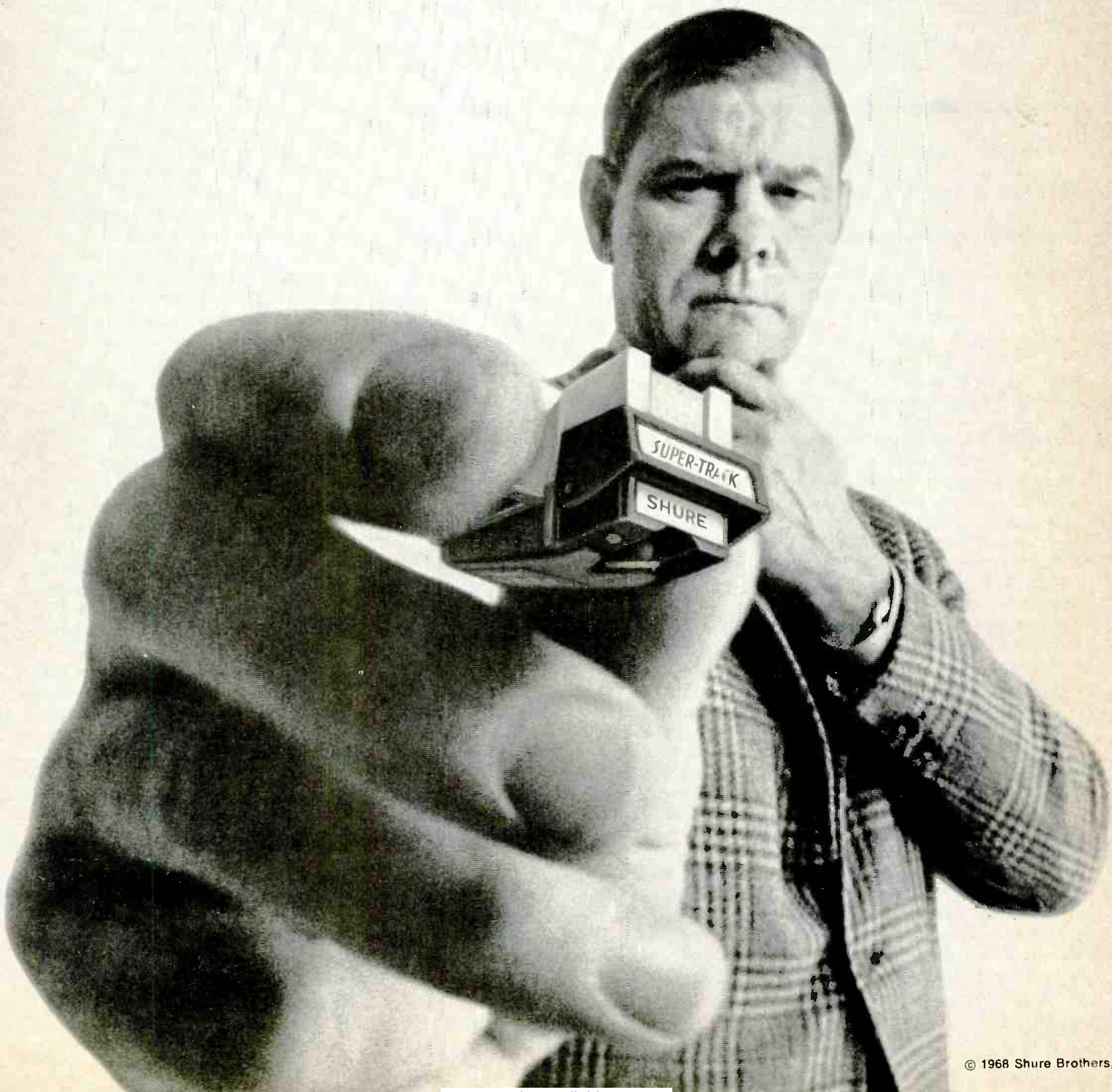
When each picture is complete, it is presented on the TV screen, all at once (as the eye sees it). Meanwhile, behind

**Fig. 1**—The "Phonovid" still-picture/sound system is shown at left, playing a special LP record on a Garrard automatic turntable. At right is Sony's magnetic-type video disc, which records and plays back still pictures in black-and-white and color, as well as sound.

# The cartridge looms large for a simple reason:

It is the point of contact between the entire hi-fi system and the recording. What happens at the tip of its tiny stylus determines what will happen in all those big and impressive components that are so obvious to the eye and, in the aggregate, so apparent to the pocketbook. Worldwide, experts and critics have hailed the discovery of Trackability as *the* definitive measurement of cartridge performance. When evaluated against this measurement, the superb **Shure V-15 Type II Super Track** stands alone. Shure Brothers, Inc., 222 Hartrey Ave., Evanston, Illinois 60204

The analog-computer-designed Shure V-15 Type II Super-Trackability cartridge maintains contact between the stylus and record groove at tracking forces from ¾ to 1½ grams throughout and beyond the audible spectrum (20-25,000 Hz). Independent critics say it will make all of your records, stereo and mono, sound better and last longer. Tracks 18 cm/sec. and up at 400 Hz; tracks 26 cm/sec. and up at 5,000 Hz; tracks 18 cm/sec. and up at 10,000 Hz. This minimum trackability is well above the theoretical limits of cutting velocities found in quality records. $67.50.



the electronic scene, another picture is being slowly drawn by the scanning beam, actuated by the record groove. Takes that full six seconds to do it.

How do you store up picture info? Via one of those tricky memory-type storage tubes. Phonovid uses two tubes, one to collect the slow playback info and the other to accept the complete fast-scanned image when the first tube is ready. There's a slight degradation, I gather, as this transfer system operates. But it isn't of serious proportions. "No problem," the engineers said at the demo. And that's the way it looked to me. Surprisingly clear, steady pictures. Sharp enough to read print and diagrams of considerable complexity. After all, how sharp is normal commercial TV? Not very.

Let me be slightly more specific, as per the notes I took at the time. The basic nominal bandwidth for the picture is, if my handwriting is correct, 10 kHz, operating via a reconstituted side band, laid out from 5 kHz upwards on the LP record. The sound track uses the lower portion of the spectrum. Not exactly hi-fi sound—but I found it a lot better than the usual dreadful sound track we hear via normal 16-mm sound film. Actually (my notes say), the system goes a bit wider in practice, up as high as 18½ kHz, giving a maximum bandwidth for the picture of some 13-plus kHz. That little extra probably helps. Given that much bandwidth, and the standard LP playing speed and playback equipment, you can come up with a good picture at the six-second rate.

The whole thing plays as long as a normal LP plays—which is a long time. Maximum is, perhaps, 30 minutes a side.

Now — for some urgent questions. You've already been asking them, I am sure.

What happens when you take the phono stylus out of the groove in mid-play? Really zany! The last picture just stays on the screen. Sits there, unmoving, unfading.

When you put your stylus back in the groove, things start going again, though there may briefly be a split image, half of one picture and half of another. (Not more than the six seconds maximum, of course.) A very odd effect! You can stop anywhere and hold.

The sound track, of course, reacts exactly as on a normal LP record. The voice stops, then starts in the middle of a word. But the picture "hangs over."

Now the big one—how about *mechanical distortions* due to the record and stylus? What do you see when there is LP surface noise? Hiss, pops, cracks, clicks? And, even worse, what about wows, warps, ripples, off-center grooving? There really ought to be unbearable degradation, yes? All sorts of snow, flashings, picture distortions, all the horrid TV "static" effects? An LP record, strictly mass produced, isn't that "clean." And, remember, no special treatment.

These engineers were clever, I tell you. They had this figured out from the word go. They thought ahead, and solved the problem. Now how about *you* closing your eyes right here and taking a long, long minute to do the same in your head, decisively, in one major more. *Got it?*

So ingenious. *Frequency modulation.* An FM signal on the disc, containing the picture info.

There are minor distortion effects, to be sure. But not as you might have expected. Some wobbly lines. And brightness may vary across the picture in extreme cases. But it takes a large amount of plain, blatant record wow to produce any serious distortion at all. Surface imperfections do *not* play back as they would in a straight "pitch for pitch" recording. FM treats them as static. Indeed, the system is quite astonishingly efficient, I'd say. It seems almost immune to distortion. The record plays its pictures as smooth as silk. And that is a very big favorable factor in potential mass-produced cheapness. No special treatment needed.

Why use disc at all—why not tape? Aha—there we go. Back to the first premise. Tape is too limited in the sense of a mass-produced and widely-distributed medium. Tape, too, is clumsier and less convenient in a dozen crucial ways. All the old factors operate, those that have kept the disc alive and supreme all these years, in spite of tape.

**"The whole thing [video disc] plays as long as a normal LP plays."**

But wouldn't tape produce a better picture? It could — of course. Given changed parameters. But since via the frequency-modulated Phonovid system most of the mechanical disc crudities are by-passed in the reproduction, the engineers say that within desirable parameters the LP record has no trouble equalling tape quality.

Finally, how about those over-all limitations Motion is out as we know, and not even considered in Phonovid. But what of color? Color would be enormously helpful, in the very places where Phonovid might be most useful. Projected slides and film strips, after all, do provide color as well as black and white.

Well, lots of things might be useful. Like optical sharpness equal to that of an optically projected picture. TV doesn't have it, in any shape.

You can't have everything. No system does. Better sound might be nice, but the Phonovid sound is already superior to most 16-mm sound, and it is easily intelligible on speech, OK for background accompanying music, too. Stereo, in this situation, would be a frill but not an important factor.

As for color, it might be managed (the engineers suggested) by two channels of info; but they had not gone into this possibility. Cross talk on the stereo disc would present serious problems, they said, for one thing .

I'll have to admit that the lack of color does seem to me the only really vital limitation in the Phonovid set-up. Here's where Sony's "magnetic" disc pulls ahead. The record-shaped video disc announced by Sony back in '66 was reported to play color still pictures on TV. In addition, the video disc can be recorded upon. This video concept requires an altogether different disc and playback/record machine, unlike the Phonovid, which in the photo here is shown using a Garrard automatic turntable.

Then there are video tape recorders. Why settle for "stills" when you can have motion pictures, in color or black and white? It's just that these devices require, like the Sony video disc, a new basic machine, compared to the familiar record turntable and LP record. So let's wait and see if this particular project ever gets out of the labs and onto the turntable. I myself think that Phonovid has something, even in face of all the complex competition in the picture world due to its mass-produced simplicity, plus the fact that I've already got a record turntable, am partial to using the medium of record discs, and own a TV set. Æ

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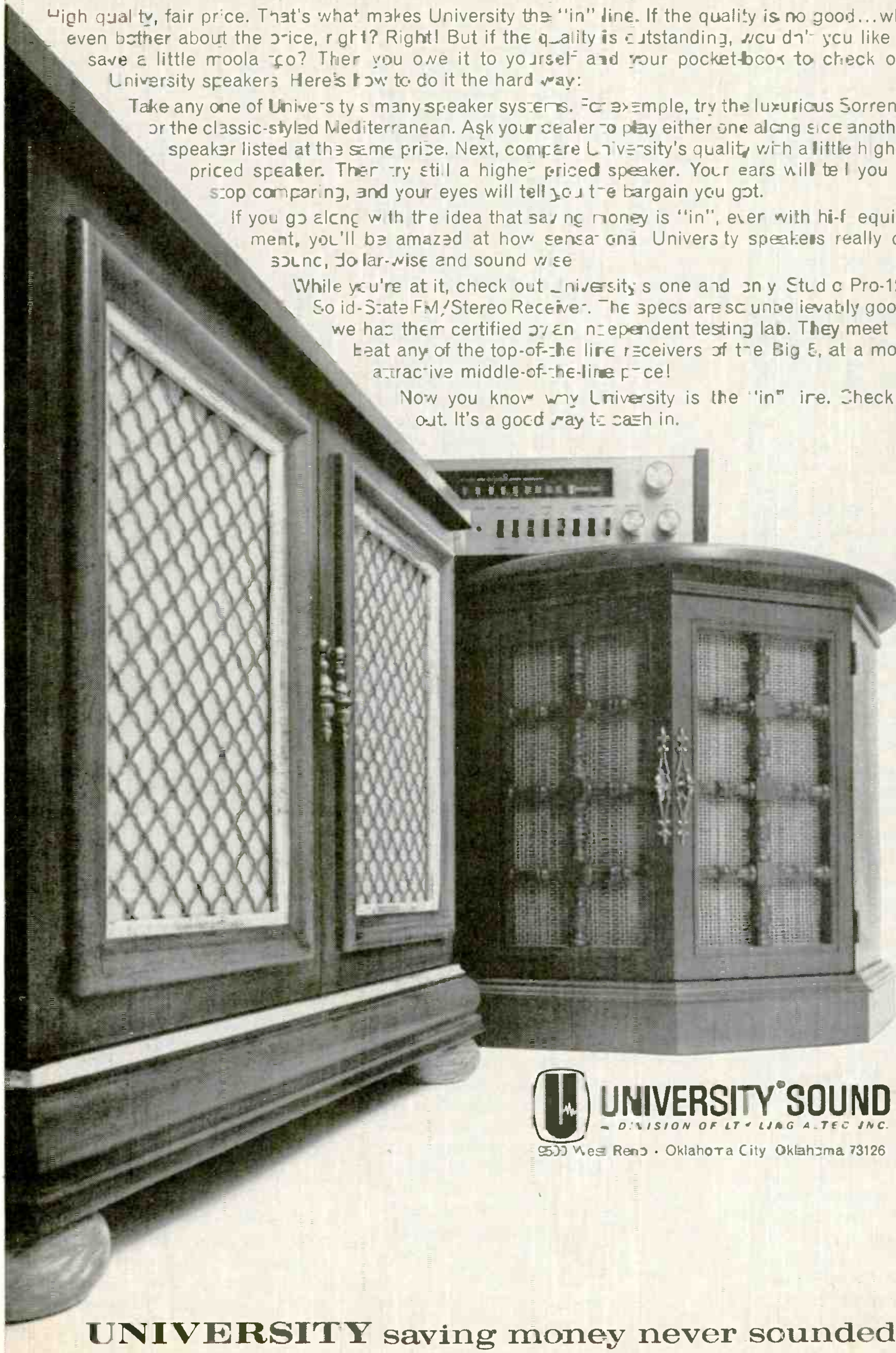

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# Tape Guide

HERMAN BURSTEIN

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

## Compatibility of equalization curves

*Q. The European CCIR curve for 19 cm/sec (7.5 ips) now sets the standard playback turnover at 70 microseconds. But one cannot tell how many and how far the manufacturers in Europe have abided by this norm or the DIN norm, which was 70 µs before the new CCIR norm was set. Before I came to live in Europe I used to curse the North American way of formulating norms left and right, to eventually try some other system, while the Europeans seemed to be so fortunate in having a body with enough authority to enforce standards. Well, it seems that chaos is here (Europe) as well. Added to the various CCIR and DIN norms, all European tape recorders usually include NAB playback; the result is that they often use NAB for recording as well.*—Andre Thiverge, Geneve, Suisse.

A. The fact that the CCIR curve now has a turnover of 70 µs, compared with 50 µs for the NAB (and RIAA) tape playback characteristics, indicates that the difference between the two characteristics is close to insignificant. The maximum difference between the two is not quite 3 dB. When one takes into account the departures from flat response deliberately introduced by engineers and others in recording, the departures from flat response of reproducing equipment, and the tolerances of the nominal playback curves, a difference of under 3 dB between two equalization characteristics approaches insignificance.

## Interpreting specifications

*Q. In the Abajian-Jones article in the October 1964 issue of* AUDIO *on building a tape amplifier, the authors state that the signal-to-noise ratio is 52 dB; that total harmonic distortion is about 1% when a signal of 1,000 Hz is recorded at the 0 VU level; and that IM distortion is then about 7%. I would greatly appreciate your remarks on these specifications with regard to how they compare with those of high-quality professional equipment.* — A. J. Steen, Los Angles, Cal.

A. I presume that for S/N measurement that Abajian-Jones are using 1% harmonic distortion, corresponding to 0 VU in recording, as their reference level. In this case the specifications for their amplifier are excellent, comparable with top professional equipment. If the S/N ratio is 52 dB at 1% distortion, it is about 6 to 8 dB higher at 3% distortion, which is a frequently specified reference level for high-quality machines. Thus the amplifier has an S/N ratio of about 58-60 dB, which puts it in a rather rare class.

Don't let the 7% IM distortion at 0 VU disturb you. This is quite normal when the tape deck and the tape get into the act. Because of the high IM figures, very little is publicly said about tape recorders' IM distortion. Only harmonic distortion is mentioned, and manufacturers don't even like to say too much about it. At the same time, keep in mind that most of the time you are recording well below 0 VU, and there is an appreciable drop below 3% harmonic and 7% IM distortion at reduced level.

## Sssssshhh

*Q. My new *** tape recorder has an annoying noise, best described as ssssssssssshhhhh, when in the playback mode. The noise is controllable by the machine's volume control, and can be almost eliminated by turning the treble control of my audio amplifier all the way down. The noise is of the same magnitude on each channel, and occurs whether or not the tape transport is in motion. If I remove all inputs from the tape recorder, the noise remains. The noise ceases when I disconnect the recorder's output cables from my audio amplifier. The noise comes through on quiet passages of prerecorded tapes. With no program material and with the tape machine's volume control at maximum, it's eggs frying at their worst. I have an older tape machine that displays less noise, but also does not have as good high frequency response as the new one. I recall several instances in your column where you told your readers that the *** tape recorder is known for its poor signal-to-noise ratio. Is my machine one of those?*—Fred J. Petzinger, Jr., Portsmouth, Va.

A. Your tape recorder is not one of those I had in mind when referring to a poor signal-to-noise ratio. Inasmuch as your new unit has a more ambitious treble response than the old one, somewhat more hiss is to be expected from the new one. In the absence of my having direct access to your new machine, or of your supplying S/N measurements, it is difficult to say whether you are getting *excessive* hiss. The answer depends partly on how loud you play your tapes; that is, on how far you advance the gain control of your audio amplifier. If you play at quite a loud level, it is normal to hear some hiss from the tape playback amplifier.

If hiss is indeed excessive, the cause might lie in noisy components in the first stage. Or it might lie in improper playback equalization. Do prerecorded tapes have an overbright sound If so, this suggests faulty playback equalization—not enough treble cut.

Another question: Do you get excessive hiss at all tape speeds? Or only at lower speeds, such as 3.75 and 1.875 ips? At lower speeds, playback equalization involves less bass boost—that is, less treble cut—than at 7.5 ips. Therefore more hiss will be apparent from the electronics at the lower speeds.

## Semantics

*Q. Does a tape deck only play tapes? Or can you record with one? Would I be better off to get an all-purpose tape recorder?*—Russell E. Webb, APO San Francisco, Cal.

A. The term tape deck is usually understood to signify a tape transport plus preamplifier electronics for playback and recording. If the electronics are confined to playback, the unit is called a playback deck. If there are no electronics, the unit is called a tape transport. A tape deck does not include power amplifiers and speakers; a unit that includes these items is called an integrated tape recorder. However, the terms tape deck and tape recorder are often used interchangeably; and the term tape recorder also is often used to designate an integrated machine. Thus there is, unfortunately, some confusion about the meaning of terms.

By an all-purpose tape recorder I assume you mean one with its own power amplifier(s) and speaker(s). This, as I have said, is also called an integrated machine. It can be advantageous if you want portability. For example, you might want to record something away from the rest of your audio system, and you might want to listen to the playback at the recording site to make sure you have what you want on the tape. Æ

# EDITOR'S REVIEW

## Hi-Fi and Liberalization

The political upheaval in Czechoslovakia coincides with an interesting event announced recently: the country's first international exhibition of stereo equipment, "Hi Fi Expo Praha 68."

The exhibition will feature FM stereo receivers, tape recorders, pickups, and other equipment of both Czechoslovak and foreign make. According to program information, there will be lectures by leading technical specialists and industrial designers, public listener comparison tests, a technical and advisory service for visitors, and even a corner for women. Interest in high-fidelity sound is increasing in Czechoslovakia, as indicated by the formation this year of a Czechoslovak Hi-Fi Club, an independent organization that now numbers over 7000 music and high-fidelity enthusiasts.

## All-Channel Radio Bill Introduced

A bill (H.R. 16523) that requires all radios to be capable of FM reception was introduced to Congress recently by Rep. Alvin O'Konski (R., Wisc.). This radio version of TV's all-channel law (television receivers must be equipped to receive both VHF and UHF broadcasts) would compel manufacturers to include both AM and FM tuners in their equipment if the bill passes.

AUDIO does *not* support this special-interest bill. It would penalize the consumer who may not wish to purchase (or may not be able to afford) two-in-one radios. Further, FM radio is exhibiting a rapid natural growth without the assistance proposed by Rep. O'Konski (who, it is reported, is the owner of a TV station and a former owner of FM radio stations).

## The Return of Carl LaFong

*Record World*, a music and record trade publication, has introduced a new weekly feature, "Notes From Underground," bylined by Carl LaFong. The column will be devoted to FM music stations that "have become a potent force in broadcasting pop music." Carl LaFong, as some readers might recall, was a pseudonym used by comedian W. C. Fields. Identity of the "new" Mr. LaFong was not revealed.

## Newport Jazz Festival

Jazz buffs note: July 4 through July 7 are the dates set for the 1968 Newport Jazz Festival. This year marks the 15th anniversary of the famous Rhode Island event. Among the artists scheduled to appear are Ray Charles and Dionne Warwick.

## I Hear Music

According to *Graphic Communications Weekly*, an engineer servicing a Vario-Klischograph electronic scanner turned in a rather strange service report. It seems that an engraving head on the machine was producing music from a local radio station. In the black-and-white position, two stations were mixed, and in the engraving-test-cut position one station was heard with clarity. Grounding the shield of the amplifier cabinet eliminated the trouble. Reading about it brought back memories of picking up broadcast music from a G.E. variable reluctance cartridge many years ago, as well as a more recent problem where a nearby, powerful FM stereo station's broadcasts were received whenever volume was raised to a normal listening level, whatever the function selected. The final solution to the latter difficulty was to move out of the area.

*A.P.S.*

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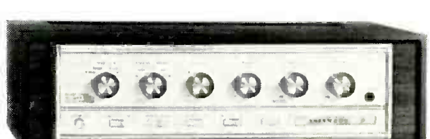
* Electronic World, Oct., 1967



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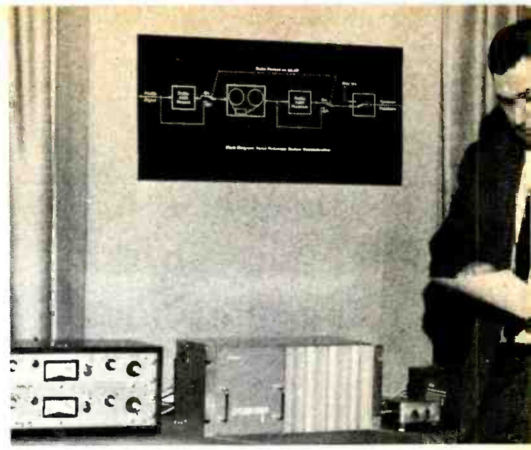


# Audio Noise Reduction: Some Practical Aspects

RAY M. DOLBY

## Part I: Operating principles and details of a professional noise-reduction system

A NEW GENERAL-PURPOSE, professional audio noise-reduction system (Dolby Laboratories model A301) was discussed last year in AUDIO Magazine.[1] The design philosophy of the system was subsequently presented in a technical paper.[2] Intended as a continuation of this discussion from a more practical point of view, the present article will examine the device itself and consider various operational aspects, especially in relation to magnetic tape recording.

The purpose of the noise-reduction system is to reduce noises—hiss, print-through, hum, as examples—that normally arise in the tape recording and playback process.

Beyond a certain point in the care taken in designing, maintaining, and operating recording equipment, any gains in signal-to-noise ratio unfortunately become increasingly difficult. Such gains may be difficult not only because of limitations imposed by the physics of the matter, but because of expense or general impracticability. A 10 dB reduction in tape hiss, for example, can in theory be obtained by the use of tracks ten times as wide. Or any desired reduction of print-through can be had at the inconvenience and expense of interleaving the layers of the recording tape with a suitable shielding material.

Clearly it is a matter of great practical significance to have a method of side-stepping these difficulties, which is the intention of the A301 system to be described.

Before going into the details of operation, it is necessary to appreciate how the system is connected into the recording chain. Referring to Fig. 1, the signal is first treated by the recording-processor half of the noise-reduction system before being recorded on tape. During playback, the signal is fed through the other half of the system, where it is restored to normal; at the same time, hiss is reduced by 10 to 15 dB (depending on the frequency), and hum, rumble, and print-through are reduced by 10 dB.

In general terms, the noise-reduction system must operate around or enclose the noise-producing element of the chain. Therefore, a requirement for use of the system is that the signal must be available for preprocessing before being fed into the noisy element and for post-processing after it emerges.

It should be appreciated that the system is not capable of separating the noise from the signal in the normal sense. All it does is pre- and post-process the signal in such a way that the signal effectively becomes less susceptible to the addition of noises. Pre-emphasis and de-emphasis and, in a somewhat different way, frequency modulation and demodulation are further instances of complementary pre- and post-processing systems which improve noise immunity.

In professional audio there are many ways in which a noise-reduction facility can be utilized, but a notable application is in the making of master tapes for high-quality phonograph records. When the original signal is put on tape in processed form it is protected from the usual sources of noise encountered in recording, dubbing, storage, and final playback. In this connection, hiss and hum are the most common noises, but print-through is undoubtedly the most serious flaw when it does occur.

### Principles of operation

The A301 system may be thought of from two points of view. First, it is a compression-and-expansion system operating in four frequency bands. Second, it is an automatic, signal-operated equalizer which continuously controls the recording and playback equalization characteristics in such a way as to improve the overall signal-to-noise ratio. From both views, the A301 is a three-dimensional signal processing system with an overall gain of unity, but with intermediate transmission properties which are functions of amplitude, frequency, and time.

Viewing the device as a compressor-expander, the main feature which distinguishes the system from previous ones along similar lines is that the signal as a whole does not pass through any variable-gain elements. Referring to Fig. 2, high-level signals pass straight through the direct path of the system (amplifiers only). Thus, they are not altered in any way whatever. By this means, the usual distortions and tracking troubles of compressor-expanders are avoided.

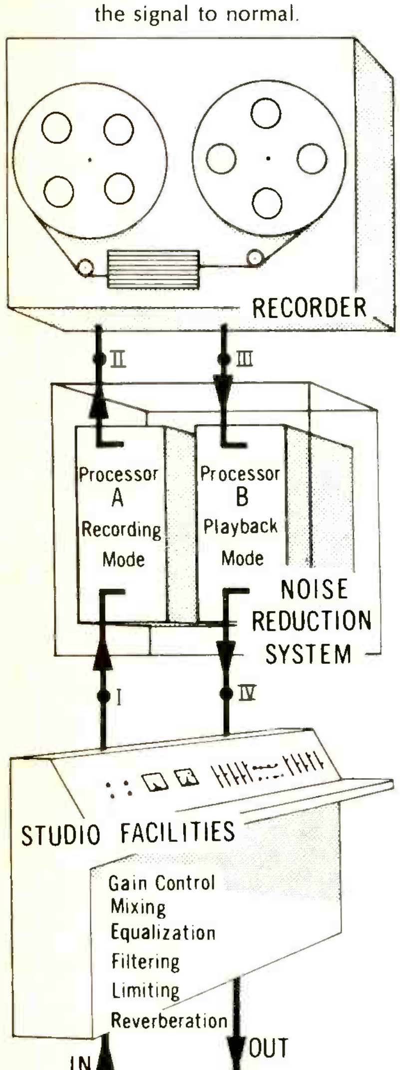
Low-level signals, which are of relevance for noise-reduction purposes, are handled in a side chain (differential network) comprising four band-splitting filters and low-level compressors. Whenever the signal amplitude is low in any band, the output from the compressor is large in comparison with the same component in the main path. The addition of the differential component to the straight-through component thereby results in a boosted output signal. The situation at high levels is that the differential component is compressed substantially; being small in comparison with the main signal, its contribution is negligible.

A complementary operation is performed during playback, with the differential component in this case being subtracted from the main signal. Since the gain of the playback unit is decreased at low levels, the desired noise-reduction effect is achieved.

An important aspect of the system is that identical differential networks are used in both the recording and playback modes. Inspection of Fig. 2 shows that, basically, an extra component is added and then it is subtracted; what is left *must* be the original signal. Insofar as correct restoration of the signal is concerned, the networks can have almost any characteristics whatever, with the proviso that they are the same.

A feature of the process, it should be noted, is that no pilot signals are used in controlling the playback operation. In effect, the signal is its

**Fig. 1**—Use of the Dolby noise-reduction system in one channel of the audio chain. The signal is processed before recording and after playback; noise is reduced and processing operations cancel out, restoring the signal to normal.



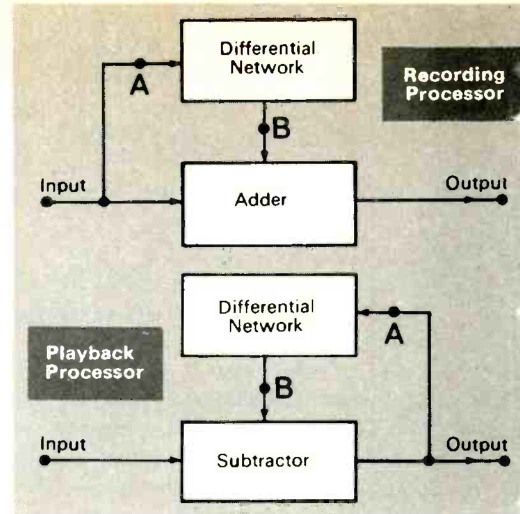

own pilot. The playback processor contains full information on the principle by which the recording unit operated; this information, together with the processed signal itself, is sufficient for the playback half to recreate the original signal.

The transfer characteristics of the two processor units are shown in Fig. 3. When the differential component, Fig. 3C, is added on a decibel (dB) basis to the input signal, the recording characteristic in Fig. 3A results. It can be seen that at very low levels the input signal is amplified, whereas at high levels the transfer characteristic essentially rejoins the input signal line. The inverse (playback) characteristic, shown in Fig. 3B, is formed by subtracting the differential component from the input signal. The result is reduced gain at low levels (noise reduction) and nominally unchanged gain at high levels.

A noise-reduction system with transfer characteristics as described above, but with only one full-frequency compressor band, would have good characteristics with regard to distortion, tracking ability, and so on, but it would suffer from poor noise-reduction properties. Full noise reduction would be obtained only at low signal levels, while at high levels the noise would have its usual value. Moreover, "swishing" and "breathing" would be produced under dynamic conditions, a familiar behavior of limiter and compressor circuits in general.

There is a fairly widespread misconception of the reason for noise-modulation effects; they are usually attributed to excessive recovery time in the limiter or compressor control circuitry. In fact, such behavior is evident even when extremely short recovery times are used. A steady-state phenomenon, the effect arises because of the inability of the signal —which usually occupies the mid-frequency portion of the spectrum— to mask low- and high-frequency noises adequately.

Fortunately, the masking effect makes it difficult or impossible for the ear to perceive noise in the *same* frequency range as the signal. By exploiting this naturally occurring noise reduction and suitably fitting compression and expansion noise reduction to it, it is feasible to provide for all normally encountered eventualities of signal and noise and to produce an overall reduction of *perceivable* noise.




The process of joining real noise reduction to apparent noise reduction must take into account the diminishing efficiency of the masking effect with increasing separation of the noise and signal frequencies. To this end, it is necessary to handle the audio spectrum in several independent frequency bands. Figure 2 shows the arrangement used in the A301 system, in which four bands are employed.

The bands are divided as follows: band 1, 80 Hz low-pass; band 2, 80 Hz-3 kHz band-pass; band 3, 3 kHz high-pass; band 4, 9 kHz high-pass. Conventional 12 dB/octave filters are used for bands 1, 3, and 4, while band 2 is designed to have a frequency and phase response which is complementary to that of the other bands.

In the recording mode, the outputs of all the bands are combined with the signal in the main path in proportions which result in a uniform low-level 10-dB boost up to about 5 kHz. Above 5 kHz, the boost rises smoothly to 15 dB at 15 kHz and then levels out. The amount of noise reduction follows the same pattern, since the frequency response in the playback processor is complementary to that in the recording processor.

For fairly low-level program material, the full amount of noise reduction is obtained in all bands. But with increasing level, the noise re-

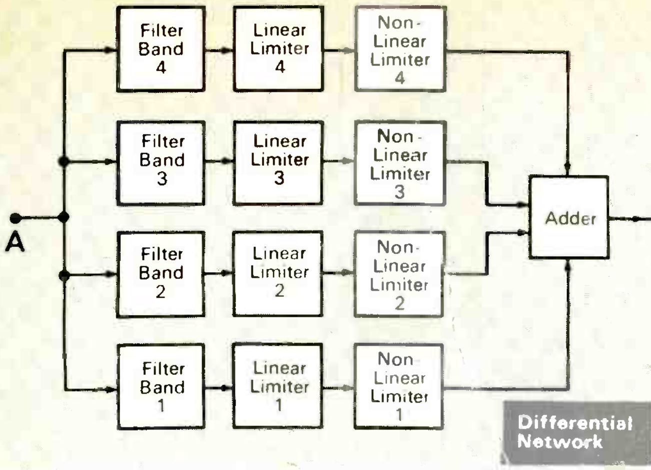



**Fig. 2** – Basic block diagram of Dolby system. During recording, a differential network adds a low-level signal to the straight-through signal. In playback, the low-level component is subtracted. The differential network (right) consists of four band-splitting filters and low-level compressors. Terminals A and B show how the network is connected.

duction in band 2 decreases progressively, whereby the masking effect then assumes control of mid-frequency noise perceptibility. Noise reduction under signal conditions therefore arises most of the time from low-level pre-emphasis, followed by complementary de-emphasis due to the actions of bands 1, 3, and 4.

Because the bands do not have sharply defined boundaries, they produce useful noise reduction outside their nominal pass-bands, a fact which has been taken into account in establishing the frequency divisions. Thus, when band 2 is paralyzed, band 1 provides noise reduction up to about 120 Hz, with band 3 being effective down to about 1.8 kHz. With signals containing fairly high-level high-frequency components, band 3 is also blocked, in which case band 4 provides noise reduction down to about 5 kHz. Band 4 is rarely blocked, except by signals such as loud cymbal crashes.

All of the bands work together, in varying degrees of momentary noise reduction, in their respective frequency ranges. The overall result is a noise level which is less (or appears to be less) than the original noise level and, equally important, is constant (or appears to be constant).

Referring to Fig. 2, a further feature of the system which should be noted is the non-linear limiter following the compressor (linear limiter) in each of the four bands. In practice, the non-linear limiter circuits are simply symmetrically biased diode clippers. Without the clippers, a tone-burst applied to the input of the system would normally cause the output to overshoot by 10 to 15 dB during the attack time (that is, the time taken for the control-signal circuitry and compressors to respond). But the amplitude of the differential component is so small in comparison with the signal in the main path, it is possible to bias the diodes in such a manner that any overshoots are confined to 2 dB with peak level inputs. Such clipping may seem to be a very dubious procedure. In actuality, however, the limiters operate linearly except with the most percussive program material. When they do operate, the clippers are inaudible because of the masking effect of the high-level transient components present in the main path.

The attack time of the system is variable in the range from about 1 to 100 ms, automatically adjusting itself to the size of the amplitude transition. For small transitions it is an advantage to use long attack times in order to minimize the generation of modulation products, but for large transitions it is clearly best to minimize the duration of the overshoot.

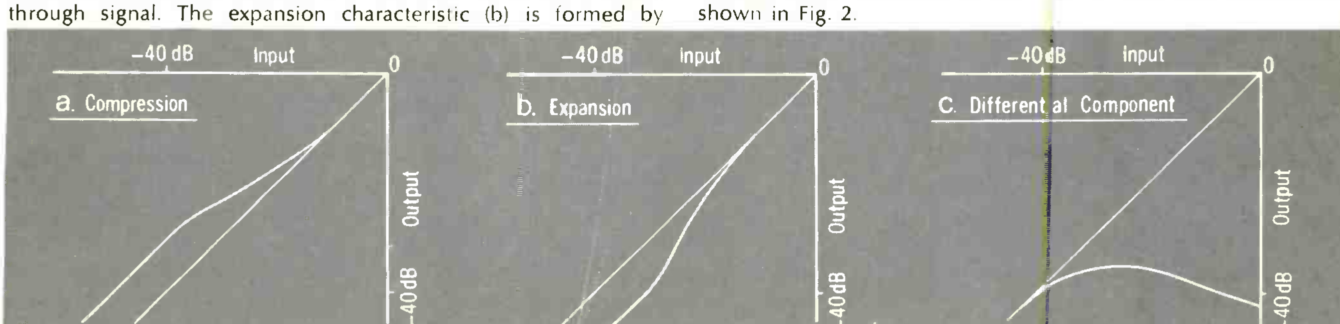
Non-linear control signal integration circuitry is similarly used to provide optimum smoothing of the rectified control signal, while minimizing the recovery time of the noise-reduction action following cessation of high-amplitude signals. Low-frequency distortion is thereby held to a fraction of a per cent at 30 Hz, and the recovery time (less than 100 ms) is short enough that no "swishing" or "breathing" effects are perceptible under even the most difficult program situations, such as "clap sticks" in a dead studio.

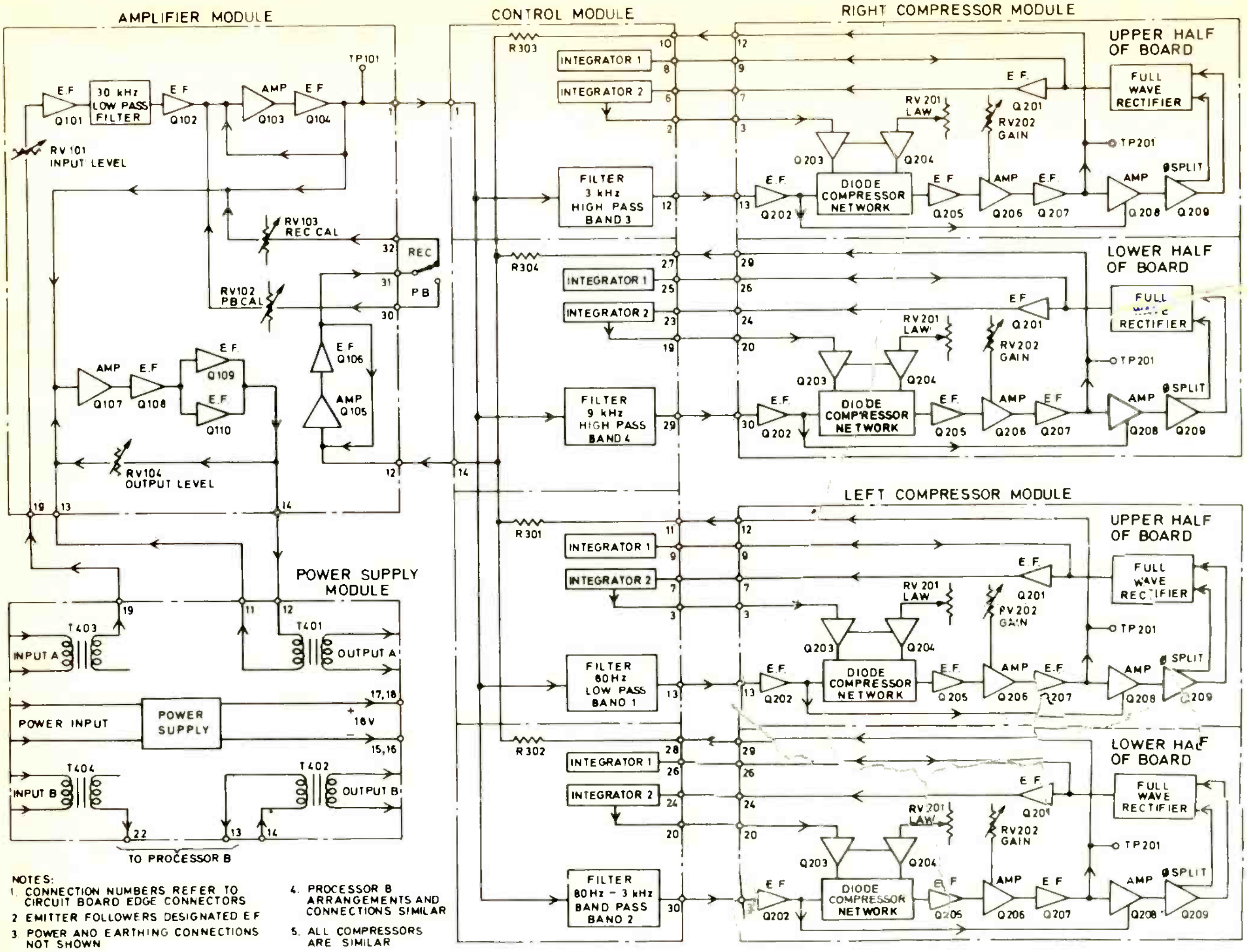
### Block diagram

Turning to the block diagram in Fig. 4, one of the two signal processors in an A301 unit is shown, together with the power supply. Each processor consists of an amplifier module, a control module, and two compressor modules (each of which contains two compressors).

The signal enters the unit through the bridging input transformer, T403 (or T404). It is fed to potentiometer RV101, which is adjusted to give a standard operating level

**Fig. 3**—Transfer characteristic curves. Compression curve (a) is produced by adding the differential component (c) to the straight-through signal. The expansion characteristic (b) is formed by subtracting the differential component (c) from the straight-through signal according to the negative-feedback configuration shown in Fig. 2.






A301 BLOCK DIAGRAM (PROCESSOR A)

**Fig. 4**—The audio- noise reduction-system consists of two identical processors, one each for left and right channels. A Processor (amplifier, control, and two compressor modules) is shown above.

within the system. After passing through the 30-kHz low-pass filter, which removes any tape recorder bias or other undesired high-frequency signals in the input, the signal is fed to the filter driver-amplifier, Q103 and Q104. The output from the amplifier is passed to the filters in the control module and also to the output amplifier, Q107-Q110.

The output amplifier itself is fairly standard, having an output impedance of 600 ohms and a clipping level a little over +18 dBm (that is, 14 dB above 0 VU on the normal +4 dBm standard). As with the input, the system output is left floating to minimize line noises.

The control module coordinates and controls the operation of the four compressors. All functions which differ from band to band are contained in this module, an arrangement allowing identical compressors to be used for all bands.

Following band-splitting, the signal is distributed to the four compressors, being fed in through the emitter follower Q202, which in turn drives the diode compressor network, a combination of two germanium and two silicon diodes.

The compressor circuit takes advantage of the fact that a diode's dynamic resistance can be controlled by the direct current flowing through it; transistors Q203 and Q204 produce a control current which determines the impedance of the diodes. The diodes form part of an attenuator network, a balanced configuration being employed to cancel the d.c. component. Because of the low signal amplitudes handled in relation to the curvature of the diode characteristics, distortion produced in the compressor is negligible.

The compression threshold in all bands is 40 dB below peak operating level, defined according to the European convention of taking the nominal 2% distortion point on magnetic tape as peak operating level. In VU terms, the threshold is 36 dB below 0 VU.

The output of the compressor is amplified by Q205 and Q206, passed through the diode clipper circuit (between Q206 and Q207), and returned to the control module through the emitter follower, Q207.

The output is also amplified further by the control-signal amplifier, Q208, and passed to the phase splitter, Q209, and full-wave rectifier circuit. The fed-forward signal from Q202 to Q208 should be noted. By suitably combining this signal with the output of the compressor, the resultant control signal produces the down-turning characteristic shown in Fig. 3C.

After being pre-integrated by the fast time-constant integrator 1 in the control module, the d.c. control signal is fed through emitter follower Q201 and back again to the control module for further integration. Integrator 2 is an RC circuit with a back-

(Continued on page 55)

# A Marantz stereo component isn't built for the mass market.

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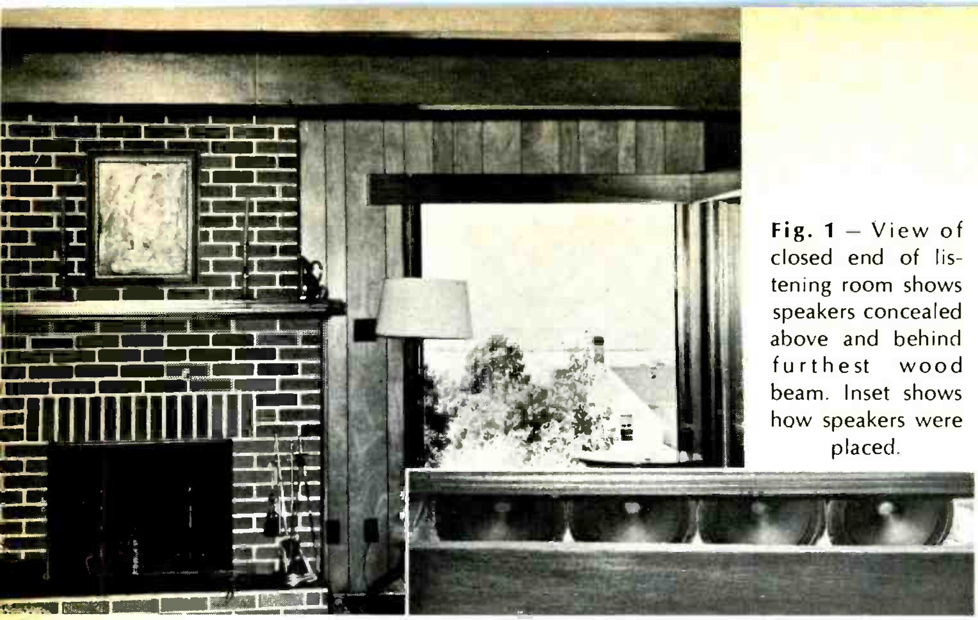


Fig. 1 — View of closed end of listening room shows speakers concealed above and behind furthest wood beam. Inset shows how speakers were placed.

# *A Custom* Home Stereo Installation

MICHAEL J. CURRY

**The author stresses room acoustics, illustrating how he determined reverberation time in his music-listening room.**

EXTENDING THE science of architectural acoustics to private residences can enhance the performance of good-quality stereophonic home music systems. The listening room is the final link in the sound chain, as most readers know, and plays a significant role in the quality of the final sound.

There are three possible approaches to solving the problem of the "right" stereophonic reproduction system; that is, integrating music-reproducing equipment and its environment. The ideal procedure would be to design an acoustically excellent listening room and then custom-design the rest of the house to fit.[1] This is obviously a procedure best suited to wealthy persons. The second and perhaps most common method is to install stereo equipment in an existing room wherever sufficient space can be found. This is a makeshift arrangement which rarely satisfies a discriminating stereo hi-fi enthusiast, though he is often forced to adopt this approach. The third method, and the best one in the author's opinion, is to design the room and the audio system together so that a harmonious accord between decor and sound is achieved. This was the procedure utilized by the author and described here.

## Construction

The basic objective of the stereo installation was to provide good listening conditions and facilities in the living room of the home, although additional speakers were provided in the master bedroom (see Fig. 3). A 70-watt transistorized stereo amplifier provides the driving power, fed from a stereo phono cartridge installed in an automatic turntable.

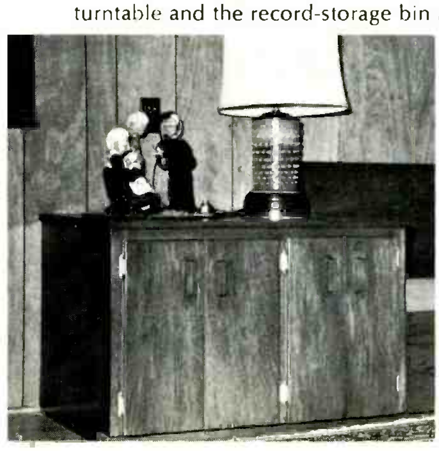
The previously-determined plans for the living room incorporated three exposed beams running across the room, as shown in Figs. 1 and 4. The beam nearest the closed end of the room was partially cut away over two 3-foot lengths to admit the mounting of eight 8-inch, wide-range speakers, four per channel, in the attic.[2] Thus, the entire attic serves as a common enclosure for both speaker systems. The radiated sound is reflected twice—once from the cutaway beam and once from the end of the room—before reaching the listener. This results in an apparent sound source across the end of the room.

A specially-designed equipment cabinet, doubling as an end-table, houses the amplifier, turntable, and controls, and provides record storage space (Fig. 2). The result is a completely built-in system which is readily accessible; yet it is pleasingly unobtrusive, not exhibiting an "added-as-an-afterthought" look of many built-in systems.

## Summary of Background Theory

Despite the considerable volume of work published in the field of architectural acoustics, there remains a good deal of controversy over the merits of precise acoustical measurements as opposed to subjective evaluation when applied to a given room[3, 4, 5, 6] Indeed, there seems to be some doubt concerning the ability of the present "state of the art" to provide meaningful and accurate descriptions of room acous-

Fig. 2—Photos of the equipment cabinet are shown with doors closed and opened. The turntable and the record-storage bin are mounted on slides for easy access.



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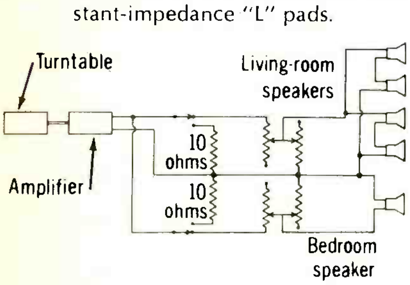
tics in terms of measurable parameters.

Certainly an acoustical engineer finds it imperative to be able to express his measurements and predictions in concise and well-defined terms. However, no amount of well-tabulated data will pacify a musician or an experienced listener to whom an audio reproduction system "doesn't sound right." Therefore, when the problem at hand is that of designing listening environments for

Fig. 3—Level and balance functions are provided for each of four speakers by constant-impedance "L" pads.




a non-technical audience, the primary criterion for success should be the realism of the sound produced by the system. This is, unfortunately, a criterion which can be applied only after the system is completed; the *a priori* designing apparently has to be done on the basis of experience and intuition.

A considerable amount of work has been done, however, to provide the designer with limits within which to work. A brief summary of this theoretical work is given in a separate section accompanying this article to provide some objective basis for evaluation of the completed installation. The basic idea here was to calculate reverberation time—the key to a room's acoustic characteristic, barring resonances and the like—using mathematical methods.

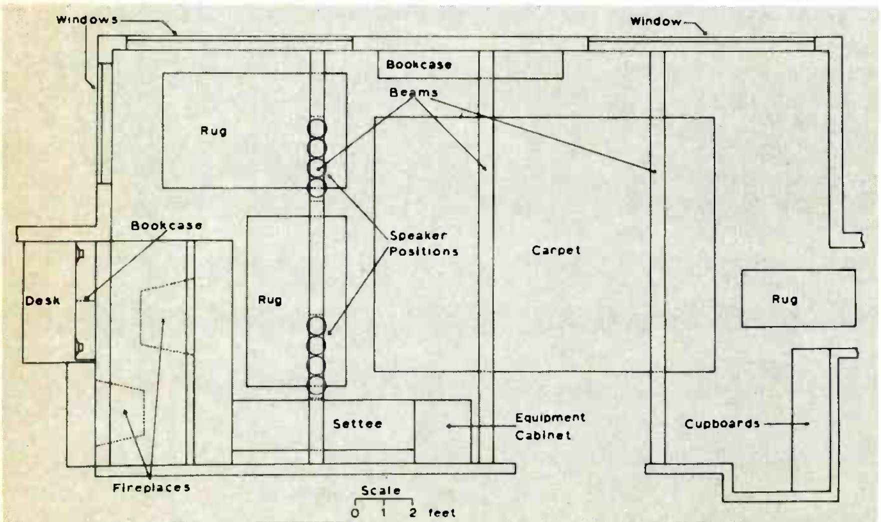
As previously stated, the real test of the stereo system was a listening test. The quality of the system's sound reproduction proved to be highly satisfying. The bass reproduction was entirely adequate, without "boom," and the highs were clear and sharp.

It should be noted that the tone and loudness controls of the amplifier provide excellent control of both tonal range and brilliance. The eight speakers were found to be capable of producing enough sound intensity to create an uncomfortable sensation without introducing noticeable distortion. In accordance with the author's preference for subjective testing, actual recorded musical selections rather than test recordings were used for the listening tests. The recordings used (all stereo) were some with which my "listeners" were reasonably familiar:

1. *The Intimate Bach*—Laurindo Almeida (Capitol SP 8582)
2. *Spanish Guitar* — Tony Mottola and Orchestra (Command RS841SD)
3. *Bossa Nova USA* — The Dave Brubeck Quartet (Columbia CS 8798)

(Continued on page 54)

Fig. 4—Sketch plan of the music-listening room illustrates placement of speakers and significant furnishings. Note that the speakers in the master bedroom, separated from the living room by the double fireplace, are mounted on the top shelf of a bookcase.




## How to Evaluate a Room's Acoustic Characteristics

THE MAIN PART of the article describes the construction and performance of a custom stereophonic system which was designed as an integral part of a new home, both acoustically and aesthetically. It suggests that a satisfactory compromise is possible between the objectives. A brief summary of the pertinent theory of acoustic room evaluation and several excellent references are provided for the benefit of those readers who wish to expand their knowledge of this fascinating subject.

Following the method of Stewart and Lindsay,[7] let us consider a room in which there is a uniform sound energy density, $E$, per unit volume. Then it can be shown that the rate of sound energy striking the room walls per unit area is

$$\text{Incident Rate} = \frac{1}{4}Eu \qquad (1)$$

where $u$ is the velocity of sound. Now, if we define the average absorption coefficient, $\bar{\alpha}$, as the fraction of the sound striking a wall which is absorbed by that wall per unit area, we see that the total absorption is given by

$$a = \bar{\alpha}S \qquad (2)$$

where $S$ is the total surface area of the room interior.

We pause momentarily at this point to mention the units of absorption. For acoustic purposes, it is assumed that an open window is a perfect absorber, with $\alpha = 1$; that is, it is assumed that none of the sound originating within a room and passing through an open window is reflected back into the room. Thus an absorber having $\alpha = 0.5$ is half as effective as an open window of the same area. The unit of absorption is the open window unit (OWU), defined as the absorption of an open window of area equal to one square foot.

From equations (1) and (2) we see that the rate of absorption of sound by the walls of the room is

$$\text{Absorption Rate} = \frac{1}{4}Eu\bar{\alpha}S \qquad (3)$$

The rate of production of sound in the room is assumed to be a constant, $A$. Then the rate of increase of sound energy in the room is equal to the rate of production less the rate of absorption. This rate of increase per unit volume is written as $dE/dt$. Then the rate of increase of sound energy for the whole room is $V\,dE/dt$. Writing the data of this paragraph in mathematical terms, we obtain the equation

$$V\,dE/dt = A - \frac{1}{4}\,Eu\bar{\alpha}S \qquad (4)$$

This equation can be solved, using standard techniques,[8] to give the energy density, $E$, as

$$E = \frac{4A}{u\bar{\alpha}S}\left[1 - e^{-\left(\frac{u\bar{\alpha}S}{4V}\right)t}\right] \qquad (5)$$

where $t$ is the time referred to the commencement of the production of sound and $e$ is the base of the Naperian logarithms (e = about 2.72).

If the production of sound is suddenly stopped, the sound energy density in the room will decrease according to the relation

$$E = E_{max}\,e^{-\left(\frac{u\bar{\alpha}S}{4V}\right)t} \qquad (6)$$

Let us now define $T$ as the time necessary for this energy density to decay to one millionth of its original value, $E_{max}$. That is,

$$E_1 = \frac{E_{max}}{1{,}000{,}000} = E_{max}\,e^{-\left(\frac{u\bar{\alpha}S}{4V}\right)T} \qquad (7)$$

Equation (7) can now be arranged in logarithmic form. Evaluating the logarithms and substituting the value for the velocity of sound in air we arrive at the expression

$$T = 0.049\,\frac{V}{\bar{\alpha}S} = 0.049\,\frac{V}{a} \qquad (8)$$

where the volume $V$ is expressed in cubic feet and the absorption $a$ in OWU.

Equation (8) is the reverberation equation, first determined experimentally by W. C. Sabine at Harvard University around 1900, and the quantity $T$ is called the reverberation time. This equation has, however, been found to be inaccurate, especially for "dead" rooms with high values of $\bar{\alpha}$. An improved formula, presented by Eyring[9], is

$$T = \frac{0.049\,V}{-S\ln(1-\bar{\alpha})} = \frac{0.049\,V}{a'} \qquad (9)$$

In view of the difference between equations (8) and (9) and as a matter of convenience, the total absorption has been redefined as

$$a' = -S\ln(1-\bar{\alpha}) = -2.30S\log(1-\bar{\alpha}) \qquad (10)$$

When $S$ is expressed in square feet, the unit of $a'$ is the Sabin. (Note that this unit corresponds to but is not exactly equal to the OWU.)

The reverberation time is the parameter most often used to determine the acoustic quality of a room. The optimum value for $T$, depending upon personal preference and the use for which the room is intended, has been expressed in several published works. Stephens and Bate[10] experimentally developed the relation

$$T_{opt} = (0.0036V^{\frac{1}{3}} + 0.107)r \qquad (11)$$

where $V$ is the room volume in cubic feet and $r$ has the values 4, 5, and 6 for speech, orchestration, and choral music respectively. The Radiotron Designers' Handbook[11] lists optimum reverberation times determined at 1000 Hz for living rooms, while Beranek[12] presents in graphical form optimum reverberation times as compiled from the literature and from experience. Values for $T_{opt}$ are also given by Knudsen[13]. The considerable variations in the optimum values suggested by these various workers render the choice of a $T_{opt}$ figure difficult; however, a basis for weighting the given values for averaging purposes has been provided by Knudsen[14], who observed that "over the period [from 1928 to 1954] there has been an unmistakable trend toward shorter reverberation times in nearly all types of rooms." Thus we are justified in attaching more significance to more recently published values of $T_{opt}$.

Equations (8) and (9) tell us that, for a given room with a fixed volume $V$, the reverberation time is dependent upon $\bar{\alpha}$, a parameter which can be varied by changing the substance of the interior of the room and which can be calculated by summing the contributions to the total absorption by the various components of the walls and furnishings[15].

$$\bar{\alpha}S = \sum_{i=1}^{n} \alpha_i S_i \qquad (12)$$

This equation enables us to calculate $\bar{\alpha}$, provided that we know the area presented by each component of the interior and the absorption coefficient of that component. Fortunately, tables of the latter quantity are readily available,[16,17,18] so that, using equations (9), (10), and (12), we can obtain a reasonable estimation of the reverberation time of a room without resort to complicated and expensive measuring equipment.

## Final evaluation

Measurements of the room and its furnishings resulted in the following values for pertinent parameters: Volume, 2630 cu. ft.; total surface area, 1350 sq. ft. Using available tables for absorption coefficients to complement the remainder of the measured data, the total absorption was calculated as shown in Table I:

### Table I
### Room Composition and Individual Contributions to Absorption

| *Material* | *Area (sq. ft.)* | $\alpha_i$ | *a (OWU)* |
|---|---|---|---|
| Floor, wood | 163.0 | 0.03 | 4.89 |
| Floor, carpeted | 168.0 | 0.25 | 42.0 |
| Walls, brick | 87.6 | 0.03 | 2.63 |
| Walls, wood panelled | 445.0 | 0.20 | 89.0 |
| Ceiling, plaster | 328.0 | 0.05 | 16.4 |
| Wood, solid | 21.1 | 0.05 | 1.05 |
| Heavy Cushion | 10.0 | 0.7 | 7.0 |
| Glass | 73.6 | 0.03 | 2.21 |
| "Half-open" (archways) | 58.6 | 0.5* | 29.3 |
| Chairs, upholstered | (4 chairs @ 3.25 OWU each) | | 13.0 |

**Total Absorption 207.5 OWU**

The remaining quantities of interest are easily calculated:

Average Absorption Coefficient:

$$\bar{\alpha} = \frac{a}{S} = 0.15$$

Reverberation Time:

$$T = 0.049\,\frac{V}{a'} = 0.58 \text{ sec.}$$

Optimum Reverberation Time:

$T_{opt}$ = about 0.6 sec.

*No mention was found in literature of a method of dealing with open archways. In this case, the area of the archways is less than 5% of the total surface area of the room, and the adjoining rooms are quite spacious. The error introduced, therefore, by assuming for these archways the value $\alpha = 0.5$, is considered to be no greater than 0.04 sec.


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Fig. 1—Dolly-builder's stereo setup (see text for component identification).

# HELLO (*Equipment*) DOLLY

JAMES P. HOLM

**Tape recordist puts equipment on wheels to simplify transporting gear from one location to another**

---

"LIVE" MAGNETIC TAPE recording is the shortest route to a high-fidelity signal. A first (or master) tape is free of several generations of loudness limiting, transfer distortions, and equalization. But the most difficult part of amateur live recording is the transportation of the equipment to the amateur live performers. This usually involves many trips, sometimes of several blocks each, between parked car and recording site; just one trip might require a safari of bearers.

One day it occurred to me that things move easier when you put wheels under them. A first thought was to simply put wheels on my sixty-pound tape machine. The next logical step was deciding to put wheels under *all* of my recording equipment. The resulting equipment dolly carries my Magnecord tape deck, among other things, and therefore was named the "Maggie-Dolly."

The Maggie-Dolly has space for eight 10½-inch reels of tape, eight microphones, two Shure M-68 mixers, and a monitor amplifier. It is currently equipped with two PML EC-71 FET condenser microphones and two EV 655-C omni-dynamics. The space behind the mixers is open for easy access to the mixer microphone receptacles. Next to the mixers is a small stereo amplifier which drives the Koss Pro-4 earphones. This amplifier also has a silicon-control-rectifier (SCR) peak detector I dreamed up to reduce recording distortion on material that has an abnormally low rms (VU-meter reading) characteristic. The peak detector reads both channels simultaneously, and its alarm light is mounted in place of the original pilot light.

Fig. 2—Front view of the dolly with drawers partially open is shown at left. A rear view, with a microphone drawer on the floor, is at right.

The large upper drawer in the Maggie-Dolly holds all the loose incidental equipment one must have when recording in the field. The smaller middle drawer is compartmentalized for microphone storage, while the small lower drawer holds the microphone power supply, 10-dB line "H" pads and line transformers for recording without the mixers.

The cable tray attaches to the dolly, as does the Maggie, with window locks. This permits removal in several seconds, and thus allows rapid breakdown of the system for transport in my car trunk. The cable tray holds enough cable to permit running two microphones out about 120 feet or four microphones out about 70 feet. The cables are wound in a figure-8 configuration around end blocks that have swinging retainers on them. This enables me to run the cable off in "fire-hose" fashion upon my usually tardy arrival at a recording session. The tray also holds four collapsible microphone stands. The elastic rope helps to hold baby booms, etc., to the tray.

The Maggie-Dolly's box is made of ⅜-in. plywood, and its frame is 1½-in. x 1½-in. x ⅛-in. angle aluminum. The box was fastened together with spiral nails and glue. The frame was welded together on a radio-frequency welder. The drawers are specially designed to prevent small items from falling out when the dolly is lying on its back during transport. The handles are swing-down motorcycle foot-rests mounted upside down. Wheels are from Sears, Roebuck & Co.

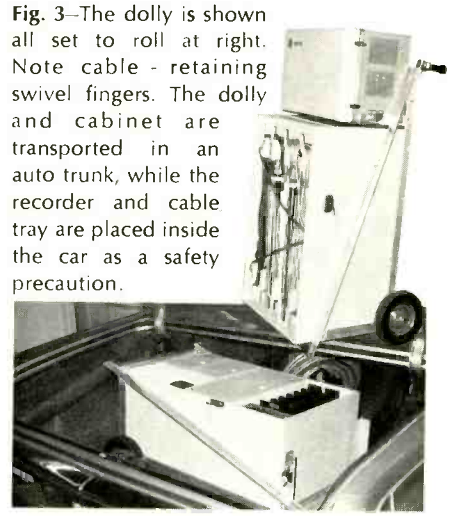
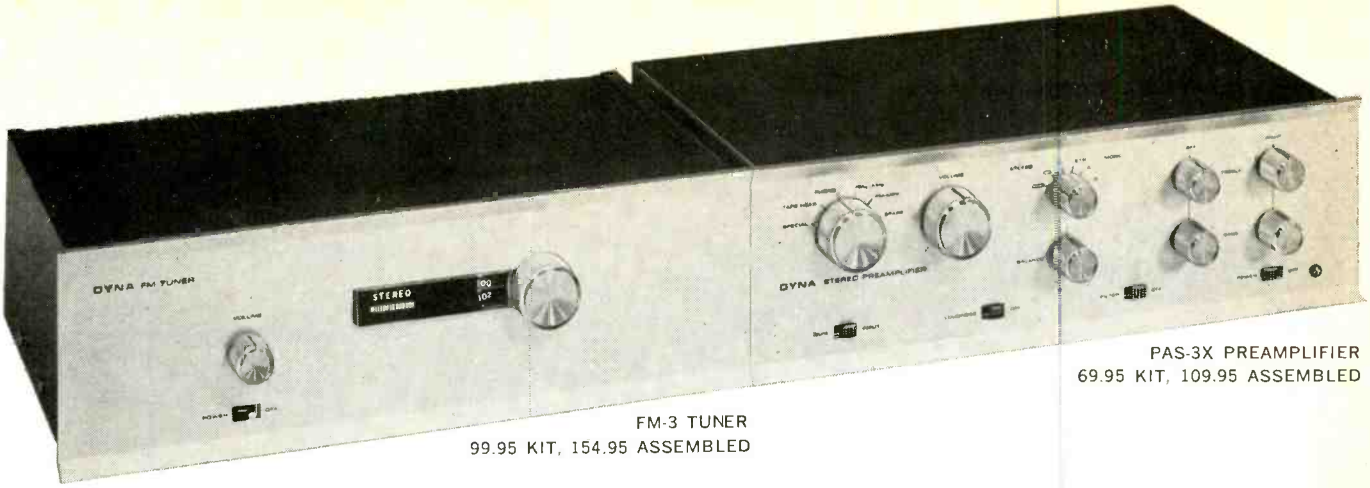


Fig. 3—The dolly is shown all set to roll at right. Note cable-retaining swivel fingers. The dolly and cabinet are transported in an auto trunk, while the recorder and cable tray are placed inside the car as a safety precaution.

My stereo system at home is situated for convenient transfer to the dolly, as shown in a photo here. The speaker enclosures are based upon a design article published in the April, 1960 issue of AUDIO. Each speaker system contains two 15-in. woofer/whizzers with a DuKane Inovac tweeter topping it off. Program sources are: (a) Dual 1019 changer with a Shure V15-II cartridge, (b) Magnecord "728" 7½- and 15-ips ½-track tape deck, (c) Viking "88" 7½- and 3¼-ips ¼-track tape deck with a hysteresis drive motor for improved pitch accuracy. The preamplifier is a Dyna PAS-3, driving a Dyna Stereo 70 basic power amplifier. Æ

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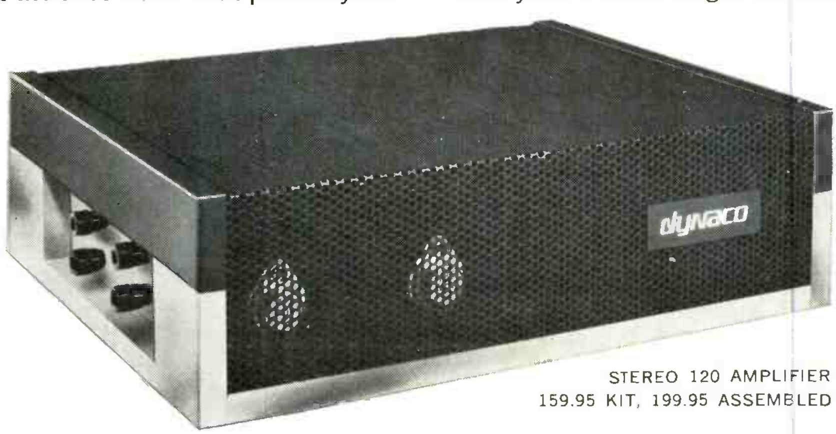


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# ABZs of FM

LEONARD FELDMAN

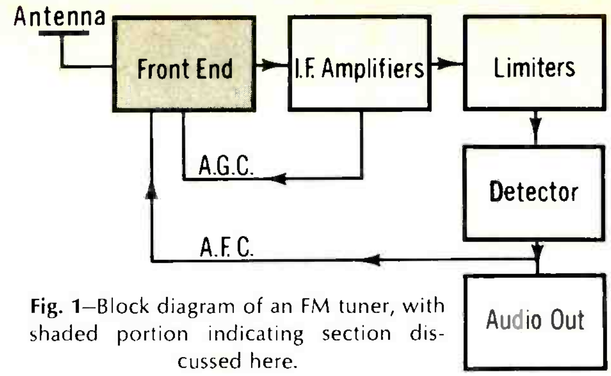



**Fig. 1**—Block diagram of an FM tuner, with shaded portion indicating section discussed here.

## R.F. Front Ends

HAVING EXAMINED some general characteristics of FM in past articles, we are now ready to begin a step-by-step analysis of the "blocks" used to make up a typical, high-quality FM tuner.

First, let's define the word "tuner," because it means different things to different people. In high-fidelity terminology, the "tuner" is all the circuitry needed to convert the received signal at the FM antenna into audio information suitable for application to an audio amplifier. "Package" or console manufacturers often refer to a "tuner" too, but they mean just the early portion of the receiver devoted to amplifying the radio frequencies and converting them to an intermediate frequency of 10.7 MHz. It is this section of a "tuner" that we call a "front-end"; and it is this section to which we shall now devote our attention.

Figure 1 is a block diagram of a typical "tuner" (by our definition). The shaded block is the one we will study first. Each month, we shall repeat this diagram, featuring a different "block" of study.

Today's tuners almost invariably employ solid-state amplifying devices in the front end, as well as in the i.f. section. More recently, many manufacturers are using Field-Effect Transistors in at least the r.f. stage of the front end. These solid-state devices more nearly approximate the performance of the highly perfected r.f. tube designs that were popular a few years ago. If this seems a bit paradoxical, one must realize that the pressures of marketing forced designers into complete transistorization a bit too soon. Only now are the solid-state devices used for front-end design catching up with some basic performances capabilities long associated with vacuum-tube performance. For this reason, we shall first examine an "old fashioned" cascode r.f. amplifier, as used in a Fisher Radio receiver some years back.

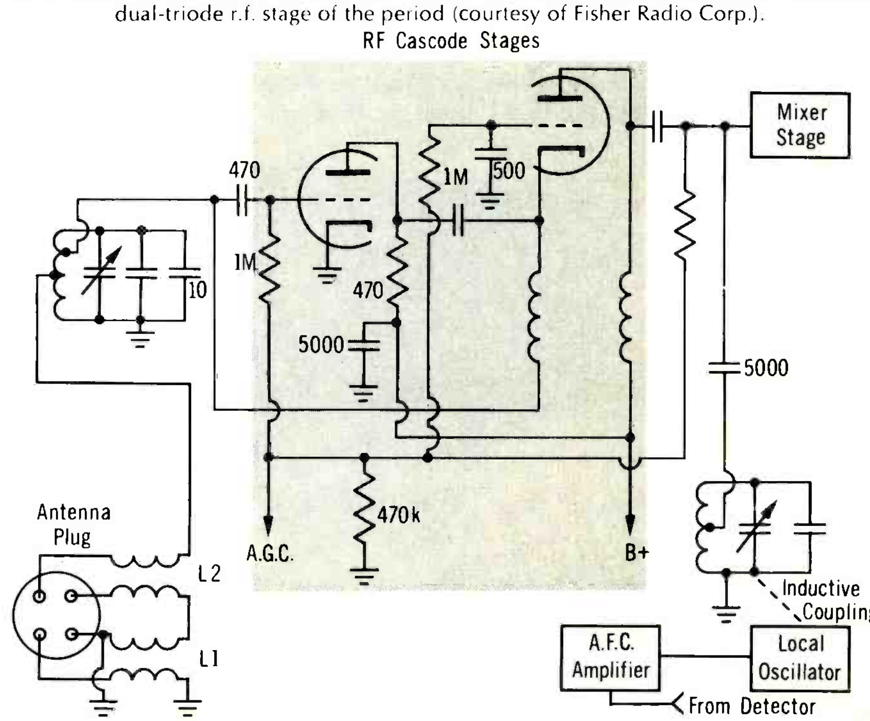
The ability of a receiver to amplify a signal is not limited by the amplification atttainable from the vacuum tubes or transistors, but rather by the noise which arises from these devices and their associated circuitry. Further, the noise developed in this first r.f. stage is actually the most significant: whatever noise voltage appears at the grid of this stage will be amplified along with the signal. The best choice for low noise (confining the discussion to tubes, for the moment) is a triode amplifier tube. Unfortunately, the gain of most triodes is less than that obtainable from pentode tubes.

The circuit shown in Fig. 2, known as a cascode amplifier, combines the gain features of a pentode with the low-noise features associated with triode operation. L1 and L2 constitutes a matching transformer arrangement known as a "balun." While most antenna transmission lines used for home FM receivers is the familiar 300-ohm twin-lead type, coaxial transmission line has been shown to be more advantageous when fighting local man-made noise, such as ignition from vehicles, etc. Coaxial transmission line sold for this purpose has an impedance of 72 ohms, and if no provision were made for impedance match, the signal lost by virtue of the mis-match to a 300-ohm receiver input might well off-set the gains resulting from the use of coaxial lines in the first place. Some high-quality sets provide inputs matched for *either* 72 or 300 ohms for this reason.

The signal from secondary L2 is applied to the first tuned circuit, which in turn connects to the control grid of the first triode section. The signal at the plate of the first triode is coupled to the *cathode* of the second triode section, while the grid of the second triode is grounded (so far as r.f. is concerned). Thus, the first stage is operated as a conventional amplifier, while the second stage is employed as a grounded grid amplifier.

The non-detailed blocks (local oscil-

**Fig. 2**—Good-quality front end from the vacuum-tube era, featuring a popular "cascode" dual-triode r.f. stage of the period (courtesy of Fisher Radio Corp.).



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lator and mixer) constitute the rest of this "front end"; operation of these blocks will be discussed in more detail at a later date. At this point, however, before we present analogous transistorized r.f. stages, it would be well to examine some of the other features of this first section of an FM tuner. For one thing, we glossed over the means of "tuning."

Tuning is generally accomplished by means of a variable air capacitor, much like those used in AM receivers. Over the years many other schemes of tuning or changing frequency have been devised. For example, coaxial variable capacitors were tried by one manufacturer some years ago. Instead of the plates meshing, as in a conventional capacitor, a coaxial capacitor consisted of a stationary cylinder into which is plunged a movable cylinder. The two are separated by a dielectric (usually a glass cylinder, onto which the outer conductive plate is heat shrunk or vacuum plated).

Permeability tuning (where the inductance rather than the capacitance of the resonant circuits is varied) in various forms has also been used in a great many designs over the past two decades. Somehow, however, the good old air-dielectric variable capacitor seems to have won out, at least insofar as high-fidelity front ends are concerned (diode tuning has of late been adopted by some manufacturers, however). Permeability tuning is still used in automotive receivers, perhaps due to space requirements and because some physical arrangements of inductance tuning are a bit more stable and less susceptible to dust and road shock.

Confining the discussion to variable capacitors, then, the next question is: "How many sections, or tuned circuits, are needed for quality performance?" As you peer underneath FM tuners, examining construction, you are likely to find some having only two-gang capacitors, others using three gangs (these are by far in the majority) and even a few employing four sections. The minimal-quality sets employing only one tuning section for r.f. frequency selection (the second gang tunes the local oscillator frequency) will, of course, have minimum selectivity. More selective sets have a three-gang capacitor for tuning the input antenna circuit, the interstage coupling circuit (as in the example of Fig. 2), and the local oscillator. Four-gang capacitors will be found in sets which employ more than one r.f. stage or in designs where the interstage coupling is accomplished by means of a double-tuned circuit.

AGC or "Automatic gain control" is

(Continued on page 57)

*More About*

# Negative Feedback

NORMAN H. CROWHURST

**PART 5**
(Conclusion)

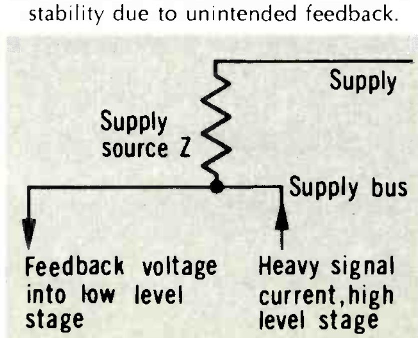
## Supply Circuits

A CAUSE OF variation in amplifier performance often overlooked is supply circuit impedance. For simplicity of calculations, supply voltages are usually considered to have zero impedance sources. But in actual amplifiers they don't. The fact that some a.c. impedance is present in supply circuits modifies feedback or provides feedback loops additional to those intended (Fig. 1).

Proper treatment can often offset this by changes in the intentional feedback, or with decoupling or isolating networks (Fig. 2). Most of these approaches assume supply circuit impedance is constant, or that voltage drop is within a predictable range. These assumptions may prove true with steady-state test conditions, using test frequencies that don't give time for the supply voltage or current demand to fluctuate, but be invalid under certain program sequence conditions.

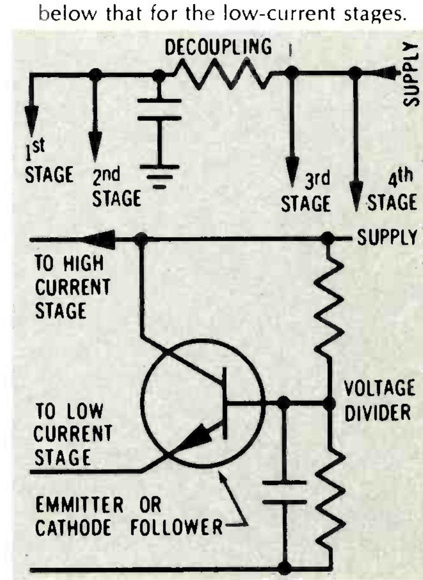
Whenever supply fluctuates, a change in supply source impedance occurs. If the source of power is battery, internal resistance changes with momentary current drain and even more with the state of charge of the battery. If the source is rectified a.c., the output impedance of the rectifier changes with momentary current handling.

Fig. 1—How supply circuits can cause instability due to unintended feedback.




Put all these effects together, and we find that feedback isn't the simple thing it looked at the beginning. Even a computer won't give you the correct answers, unless you put in all these variables. In my opinion, the better way is to start with the more important facts you know, keeping aware of the others we have mentioned, which you hope may not bother you until you've investigated the more important ones. Then if behavior does not come up to expectations, you have some ideas to work from in looking for the reasons. Applying the various pieces of information we have covered, as they prove relevant, checking possible causes to eliminate those that are not relevant, and adjusting or changing your design to overcome problems successively as they appear, will eventually care for all the possible deviations.

Fig. 2—Methods of offsetting unwanted supply-circuit coupling: (a) conventional decoupling; note that no three stages involving phase reversal should be fed from a common, undecoupled point; (b) isolation provided by emitter follower; here the high-current stage can fluctuate supply-circuit voltage, provided it doesn't dip below that for the low-current stages.




In this article, we have taken feedback apart and looked at it in a way that is a little closer to the practical circuits that use it than was the theory of previous articles. Now we have a better overall picture of the whole bag of tricks. A near-future series of articles will take some specific types of circuits and explore them. Æ

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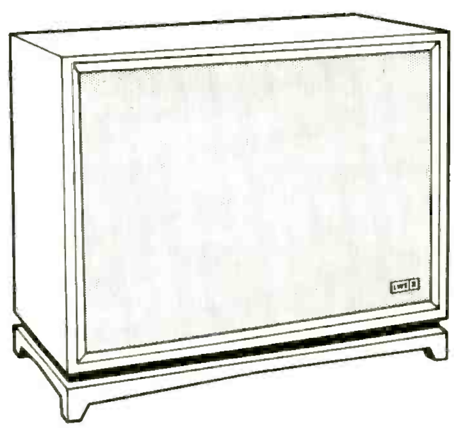


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# Equipment Profiles

*This Month:*

- Marantz Model Eighteen Stereo FM Receiver
- BSR Model 600/M44-E Automatic Turntable
- Jensen Model TF3B Speaker System

## Marantz Model Eighteen Stereo FM Receiver

Fig. 1

**MANUFACTURER'S SPECIFICATIONS – (AMPLIFIER SECTION) Power Output per channel (both channels operating): 40 watts rms at 4 and 8 ohms. Power Bandwidth (IHF): 10 Hz-30 kHz at 0.2% THD. Frequency Response: 20 Hz-20 kHz ±0.5 dB. Total Harmonic Distortion (at rated power): 0.2% max., less at lower power. IM distortion (at rated power): 0.2% max., less at lower power. High-Level Hum and Noise: –80 dB. Damping Factor: 30 min. (TUNER SECTION) FM Sensitivity (IHF usable): 2.8 $\mu$V. Total Harmonic Distortion (400 Hz, 100% mod.): 0.2% max. Multiplex Separation: 45 dB at 1000 Hz. Subcarrier Suppression: 65 dB Min. (GENERAL) Dimensions: 18¼" W x 6" H x 5¾" D. Price: $695.00.**

A Marantz FM/FM stereo *receiver!* Six-hundred-and-ninety-five dollars! And therein lies drama.

Marantz was more cautious than most manufacturers in combining on one chassis preamplifiers, power amplifiers, and a multiplex FM tuner. The company had always traveled the separate-component high road with a few other manufacturers of truly elite equipment. Here the rally was for the buyer who wanted the best, hang the cost and the problems attendant with greater space requirements of separate components.

Now Marantz has introduced an all-in-one component receiver which, at $695, rings the bell as the highest-priced receiver available at this time. With Marantz' line of separate components costing twice as much, however, the new receiver's price might be considered to be startlingly low.

### Features

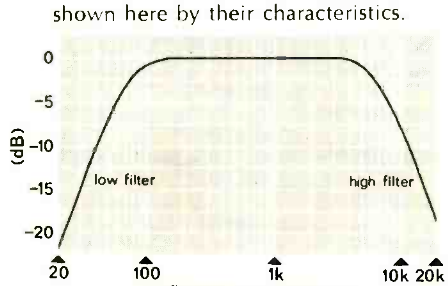
The Marantz Model Eighteen receiver is imposingly big, as you might expect it to be. It's about an inch or two more in length, width and height than most other high-power receivers.

This does not detract from its general appearance, however, since the front panel is tastefully designed. Embossed sections in black complement gold-colored surfaces at the top and bottom of the panel. The entire panel is a solid casting, hinting at the quality of construction lying behind it.

The lower half of the panel contains the usual selector switch with positions for phono, FM, tape, Aux 1 and Aux 2. Next in line are the balance control and volume control, followed by clutch-operated separate bass and treble controls, a speaker selector switch, stereo headphone jack and the power on-off toggle switch. At the extreme lower left of the panel are two additional jacks used for dubbing from one tape recorder to another—an extremely useful feature for the user who wishes to copy tapes without going to the rear of the receiver each time this dubbing is attempted.

The upper section of the panel contains the tuning dial, which is itself unique to this equipment. Instead of the usual control knob, the tuning dial consists of a horizontally mounted flywheel, the front edge of which protrudes through the front panel. This edge is knurled or serrated, so that by passing one's thumb or forefinger across this edge, tuning from one FM station to another is effected with far less effort than is required for the twisting motion of a conventional tuning knob. The dial scale itself is fully ten inches long, permitting extremely accurate fine-tuning. This demands precise calibration and alignment on the manufacturer's part so that this capability can be fulfilled. To Marantz' credit, we found that frequency calibration was never off by more than a pointer width, despite the expanded dial scale. The upper portion of the panel also incorporates a stereo light to indicate stereo FM reception. In addition, Marantz' famous oscilloscope display and its associated centering controls are located here, of which we shall have much to say shortly.

Fig. 2—Pushbutton-activated low- and high-frequency filters are unusually effective, as shown here by their characteristics.




In the interest of clean appearance and functional design, the many secondary controls have been blended into the black center strip which divides the upper and lower sections of the dress panel. These take the form of eight black push-button switches of the "push to actuate-push to release" type. The first of these buttons, when depressed, connects a second phono input pair of jacks, for the many users who have both a record changer and a manual turntable. The second button creates a monophonic or mixed L and R signal, useful for cancelling out noisy stereo FM reception. The third button effects the necessary circuit break for "tape monitoring" when used with three-headed tape decks or recorders. The next button provides for an alternate use of the 'scope display. When depressed, the scope displays combined left and right audio information; when released, the scope is used for accurate center-of-channel tuning of FM stations and many other analyses of reception quality which will be discussed later. The fifth button causes a blending of left and right channels to take place at high frequencies only. Much of the noise associated with weak stereo FM reception can be cancelled with the aid of this feature, with only a moderate reduction in apparent stereo separation.

Fig. 3—Rear panel of the Marantz Eighteen stereo FM receiver.

High- and low-frequency filters are inserted in the circuit by means of



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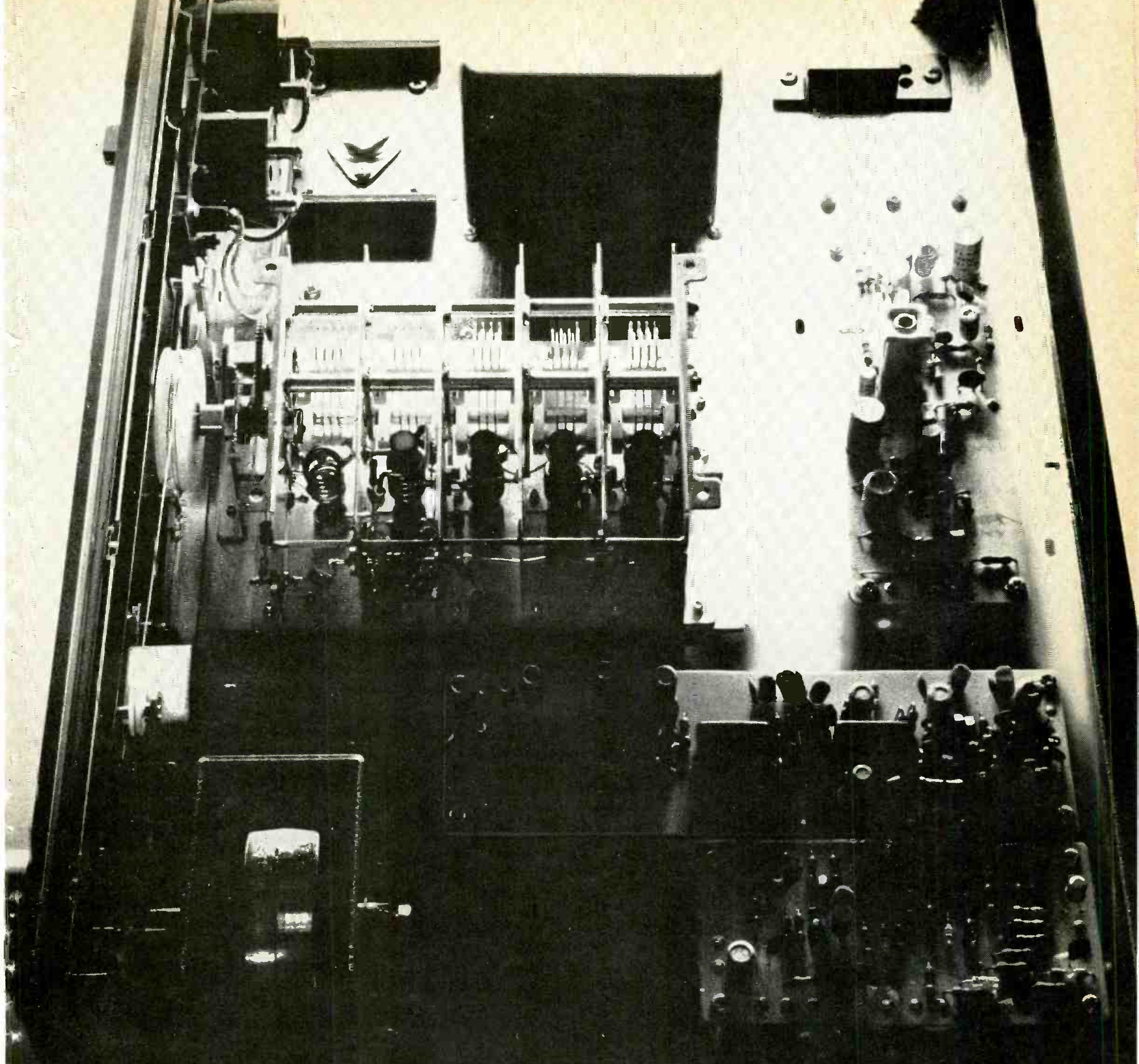

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## New Sony FET Stereo FM Tuner

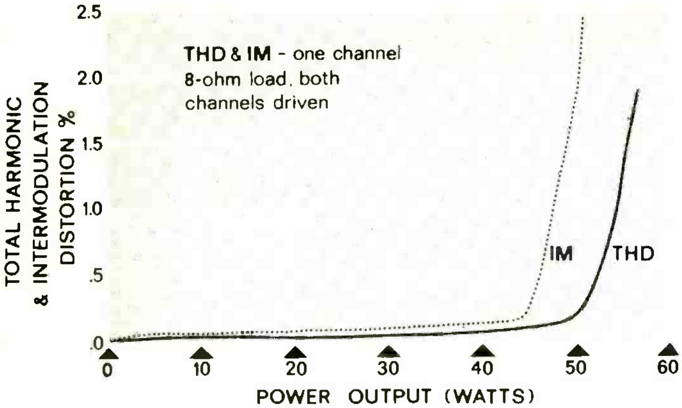



**Fig. 4**—Harmonic distortion and IM distortion of the Marantz Eighteen's amplifier section.

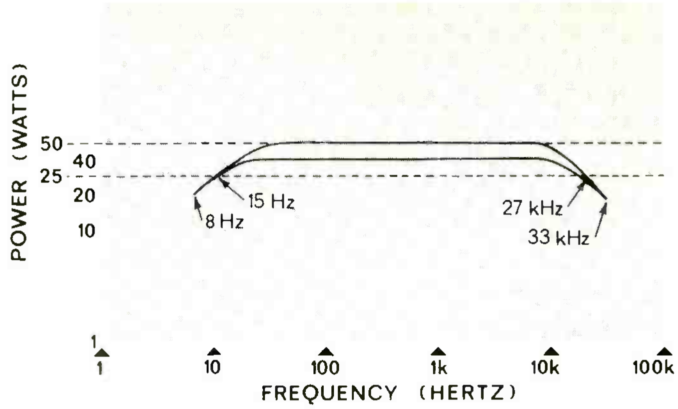



**Fig. 5**—Power bandwidth, referenced to both 40 watts and 50 watts per channel (8-ohm load).

the sixth and seventh buttons and, as can be observed in Fig. 2, they are designed with a 12-dB/octave slope, beginning at 8 kHz and 70 Hz, respectively. Unlike many so-called filters which exhibit only a 6-db/octave slope (and are therefore really nothing more than a second set of fixed tone controls), these filters are very effective in reducing rumble and high-frequency record hiss without seriously affecting overall tonal response.

The last button in this secondary control grouping defeats the interstation muting feature which is otherwise present. Normally, with the muting feature in the circuit, signal strengths of approximately 15 microvolts will overcome the muting and provide noise-free reception. But there are doubtless some DX'ers who prefer to receive distant stations even if they are noisy.

As for the rear connection panel, it contains the necessary input jacks and a pair of tape recording output jacks. These output jacks are always connected in parallel with the "Dubbing OUT" jacks on the front panel. The tape input jacks at the rear, on the other hand, are automatically disconnected when a tape recorder is connected to the "Dubbing IN" jacks at the front panel.

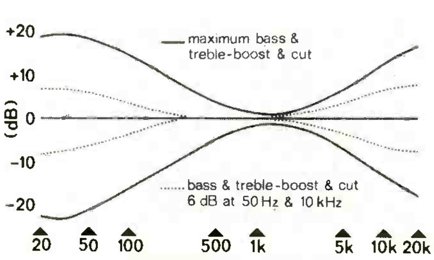
There is a single 2½-ampere fuse on the rear panel, since speaker line protection and output transistor protection is afforded by self-resetting circuit breakers. An unswitched a.c. outlet for auxiliary equipment is provided, and can be used for equipment requiring up to 500 watts. Speaker connections are made on a barrier terminal strip, as is the FM antenna. Proper matching of both 75-ohm and 300-ohm antennas is provided for by means of a balun-matching transformer. A convenient grounding terminal post completes the layout of the rear panel, which can be seen in Fig. 3.

## Performance

Measurements on the Marantz receiver indicated that specifications published by the manufacturer were very much on the conservative side. For example, the power rating which we would apply to the amplifier section would be 50 watts/channel rms with both channels operating, whereas the specs claim only 40 watts/channel. It is at 50 watts (continuous power) that we reached the incredibly low total-harmonic-distortion (THD) figure of 0.2%! IM distortion reached the 0.2% point at 45 watts. Curves of IM and THD referenced to an 8-ohm load may be examined in Fig. 4.

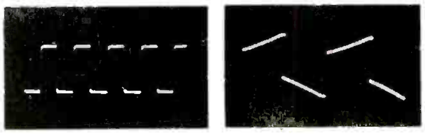
Two power bandwidth curves are shown in Fig. 5. The lower curve is referenced to 40 watts at 0.2% distortion, extending from 8 Hz to 33 kHz. Note that it exceeds the published claim of 10 Hz to 30 kHz. For consistency, we also plotted power bandwidth for a 50-watt level (which is the power rating we would assign to this amplifier) and came up with end points of 15 Hz and 27 kHz. Remarkable!

Fig. 6—Tone-control characteristics illustrate how "customized" compensation is achievable.




Tone control action is illustrated by the double set of curves shown in Fig. 6, in which the dotted curves represent partial rotation of the bass and treble controls. From these curves you can see that the variable crossover, feed-

Fig. 7—Square-wave response at 10 kHz (left) and 100 Hz.



back type of tone-control circuit is used, enabling a degree of "customized" tonal compensation not possible with less expensive "losser" circuits. As for frequency response, rather than resort to special graph paper, suffice it to say that we measured flat response from 8 Hz to 46 kHz (+0,−1 dB), again surpassing published specifications.

The FM section of the receiver proved to be an excellent match for the audio amplifier section. For example, the tuner section's harmonic distortion figure for 100% modulation at 400 Hz, which is stated as 0.2% maximum, measured *0.1%* on our sample.

FM sensitivity (IHF) read 3.0 microvolts at 98 MHz. A passive front end—that is, one that provides no amplification whatsoever—holds down the sensitivity a bit, but more than makes up for it in other areas, as will be dis-

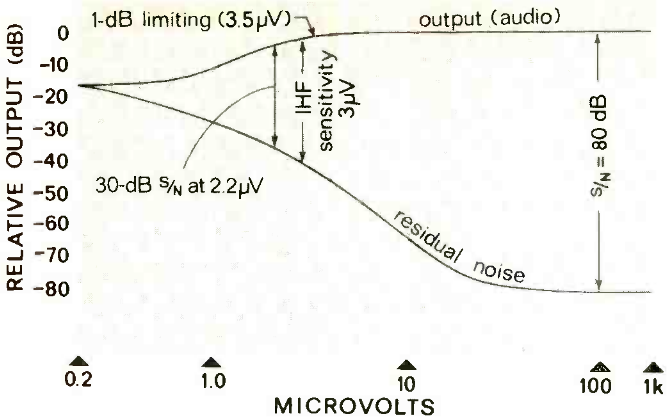



**Fig. 8**—FM quieting sensitivity of the Model Eighteen receiver with 100% modulation, 400 Hz. The 80-dB quieting figure at 100 microvolts and up is the greatest figure AUDIO has observed to date.

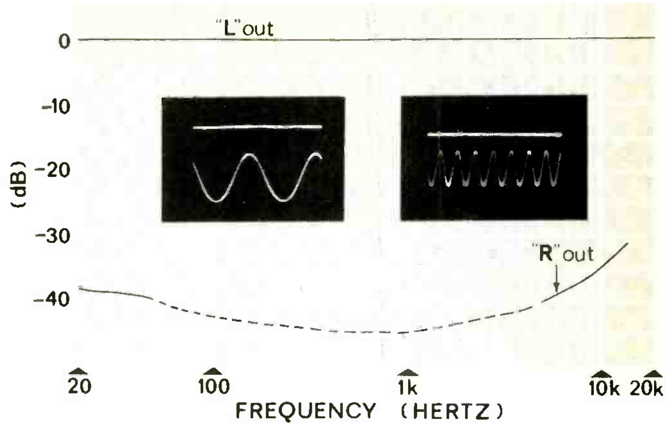



**Fig. 9**—FM stereo separation, left-channel output only (right channel is identical). See text concerning dashed line. Scope traces show separation at 1 kHz (left) and 10 kHz.

cussed later. Quickly, though, the design innovation makes it possible to receive more listenable stations than previously possible. The quieting sensitivity is plotted in Fig. 8. It ties in very well with published claims, exceeding the claims at 10 and 50 microvolts and reaching the incredible quieting figure of 80 dB at 100 microvolts and up. This is the *greatest figure of quieting we have ever observed with any FM receiver!*

Stereo FM performance and separation is plotted in Fig. 9. Only the left output is shown with respect to residual right output, since the reverse plot is so close to this one that the lines would be superimposed upon one another. The dashed-line area of the lower (separation) curve is a bit embarrassing—for us, that is. You see, our stereo FM signal simulator has a guaranteed separation capability of 40 dB. This is fully 10 dB more than is required of stations transmitting FM stereo, and we always thought it would be adequate for any equipment tests we might have to make. But here is the Marantz Eighteen, which claims separation of 45 dB MINIMUM at 1000 Hz! They probably make it or come close—but we will have to accept their word for it, since our equipment cannot confirm anything beyond 40 dB.

Invariably, when a prospective customer is confronted with a receiver retailing at around $700.00 in the face of competition in the $300, $400 or even $500 price class, he will ask, "What makes this one worth that much more?" or "Is it really that much better?" There is no "pat" answer to these questions, but here are some facts that may help you to decide for yourself.

With the exception of the Cathode ray tube ('scope tube), the Model Eighteen is an all-transistor receiver, embodying some of the same circuits and design philosophy as the Marantz Model 10B tuner, the 7T Preamplifier and the Model 15 Power Amplifier (combined cost: $1470, less cases). Its circuitry uses 73 transistors and 76 diodes, or a total of 149 solid-state devices.

Examination of the insides of this unit (see Figs. 10 and 11) discloses the use of parts such as electrolytic capacitors, toroidally wound inductors, precision resistors (often having power ratings four and five times greater than would be required in the given circuit) and even mechanical parts more often associated with the reliability and durability of military or industrial electronic equipment. The power transformer, which can be seen in the photos of Figs. 10 and 11, is larger than any we have seen since the days of high-powered tube-type power amplifiers. One half the size might easily have powered this receiver—but would it have run as cool to the touch after several hours of use as this one does?

The i.f. section and limiter section of this receiver does not use conventional i.f. transformers. To quote the Marantz manual:

"The i.f. section is a modified Butterworth-type filter configuration. The characteristics . . . are ideal in that the 200 kHz pass-band is phase linear with sharp cut-off slopes . . . assures the elimination of a major source of high-frequency distortion and loss of stereo separation . . . permits reception of adjacent channels under adverse reception conditions. This i.f. filter permits performance which is unobtainable with conventional i.f. transformer coupled circuits."

By way of illustration, a single such filter circuit is shown in Fig. 12. A glance at the number of components involved suggests the number of coils and capacitors usually found in many *entire* i.f. strips. Yet there are FOUR such interstage circuits in the Marantz i.f. strip, as distinguished from the limiter strip, which is a separate circuit

**Fig. 10**—The Marantz Eighteen receiver's outsize power transformer, seen in this top-side photo, runs exceptionally cool.

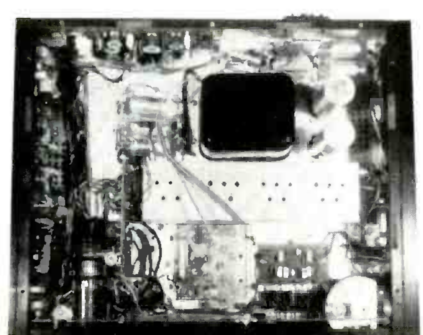
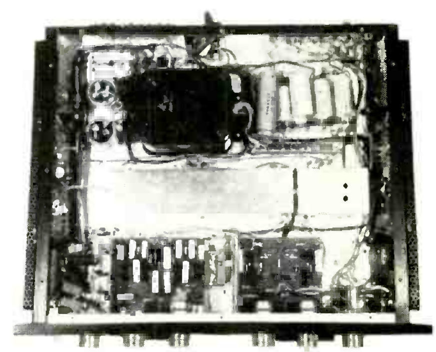


**Fig. 11**—The underside reveals high-quality components more often associated with industrial or military equipment.



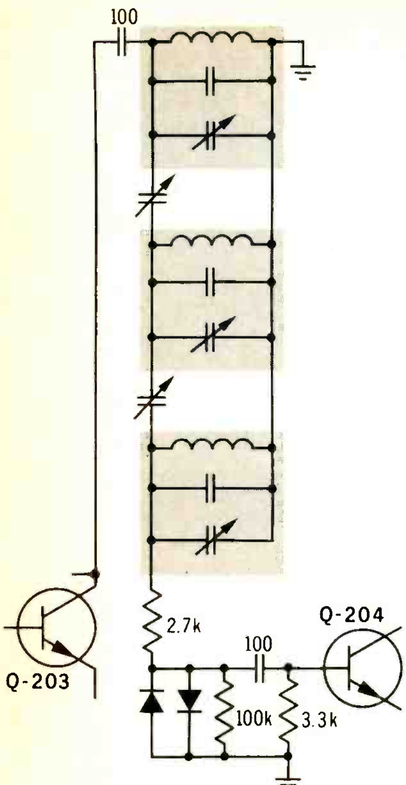



**Fig. 12**—One of several complex band-pass filters which are used instead of conventional i.f. transformers in the Marantz i.f. strip.

module containing four *additional* limiter stages!

Then, of course, we have the oscilloscope display. Is it just a "gimmick" or is it, indeed, an aid to better listening? Having had an opportunity to use it for several weeks, we can say without equivocation that it is a great aid to the serious FM listener intent upon achieving the best reception he can. The various traces of Fig. 13 tell the whole story. Trace A represents an FM station, with audio information applied, properly and centrally tuned. Had the station been detuned, the trace would appear either to the left or right of the vertical center line. (Up to this point, a "center of channel" tuning meter would do just as well.) The traces of Fig. 13, labelled B and C, show various degrees of "multipath" or signal reflections which cause a change in *amplitude* to the received signal. Such multipath reflections can seriously impair reception of stereo FM, causing distortion, decrease in separation and even momentary shifting of left and right channel information. The solution? Re-orientation of an adequate FM antenna to reduce the "multipath." But except for spotty aural detection, ***how would you ever know without this visual aid?***

The traces of Fig. 14 illustrate the visual displays that might be seen when the alternate function of the display 'scope is used—the one engaged by depressing the front-panel pushbutton described earlier. Here, left- and right-channel amplitudes, singly or in combination, are visually apparent. Note that monophonic reception is indicated by a sloping line, denoting equal left and right information. An interesting fact is that we caught a station ostensibly engaged in stereo broadcasting playing a monophonic recording on four occasions. Ordinarily, the stereo light would have contributed nothing to our knowledge since it is illuminated whenever the station's 19-kHz pilot carrier is turned on. Once, when we were sure the recording involved was only issued in stereo, we actually telephoned the broadcast station and, sure enough, someone had failed to throw the "mono-stereo" (19-kHz pilot switch) in the studio.

All right, so the Marantz Eighteen enables the user to monitor a station's proficiency or deficiencies. But how does the equipment *sound?*

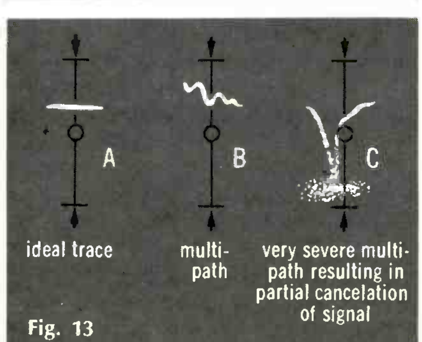
We found that the Marantz receiver provided a subtle superiority in performance over other modern receivers examined. This is not always immediately discernible, but it becomes clearly evident after awhile. For example, though 3 microvolts (IHF) usable sensitivity is not the best figure we have ever encountered in a receiver's tuner section, we were able to listen to 42 FM stations with satisfactory quieting and low enough distortion to make them truly listenable. This is four more than we have been able to receive on receivers measured heretofore. Conclusion: it takes more than just "sensitivity" to receive noise-free, distortion-free FM. Perhaps this is due to the elaborate i.f. system or the *passive* FM front end (no transistors, not even FETs; all the "gain" is accomplished at conversion or in the i.f. section—hence, no "cross modulation" problems or spurious responses attributable to non-linear characteristics of front-end amplifiers), or both.

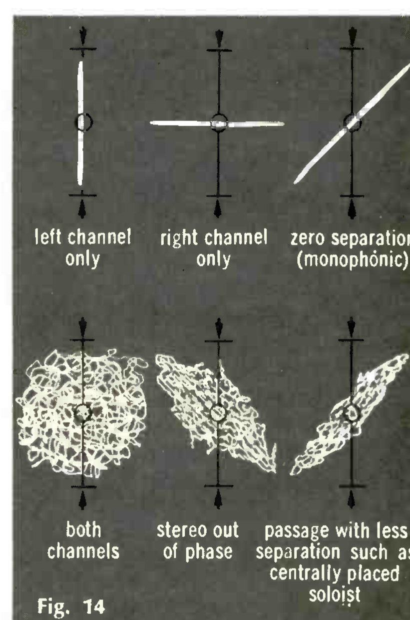
In the presence of a strong signal, the Model Eighteen imparts a cleanness of stereo in FM listening that approaches listening to master tapes. Obviously, separation, per se, is not the only criterion. When we increased the vertical gain of our scope, after photographing the separation characteristics of the stereo FM portion of the receiver at 1 kHz and 10 kHz (Fig. 8), we noted that what little "cross talk" there was (not visible in the photos because of the scale used) was not made up of second and third harmonic components, but was primarily a fundamental of the signal in the opposite channel.

Records were reproduced on the Model Eighteen with much less apparent IM distortion than we were used to hearing. Truly, it can be said that whatever IM remained was a function of the cartridge and not the preamplifier or amplifier. We tried, in vain, to tax the dynamic range of this amplifier during our LP-record auditioning, employing everything from intimate string quartet ensembles (full of pauses and quiet passages) to large orchestral works. With some of the better-quality

**Fig. 13** (below)—The Model Eighteen's built-in oscilloscope can pinpoint the spot on a dial where minimal distortion is present.

**Fig. 14** (right)—Here are some of the various audio displays which are observable on the Model Eighteen's scope.



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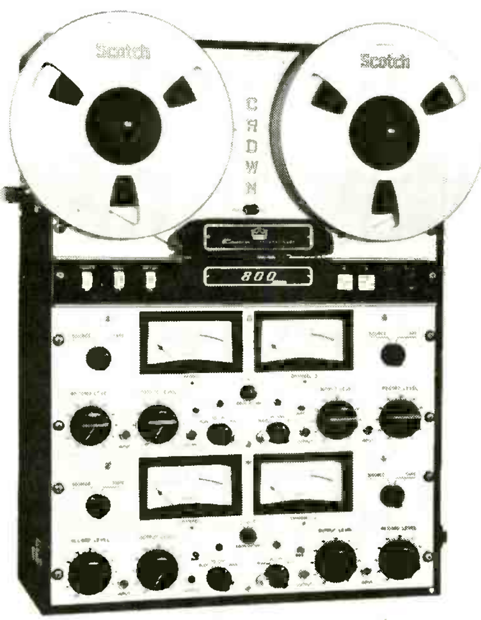


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recordings, there was the wonderful feeling of transparency that good solid-state amplifiers impart.

Based on the foregoing tests and observations, the Marantz Eighteen receiver appears to bridge the gap between separate components and today's increasingly popular receivers. It looks like a receiver, combining all electronics on one chassis, while it performs like good-quality separate components. Though it is not quite the peer of Marantz' own line of separate components, the Marantz receiver shares many of its design and long-life construction virtues. And it's half the price! So if you've aspired to own Marantz equipment in the past and could never swing the price, there's another turn at bat for you.

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### Addenda to Sony/Superscope Model 230 Stereo Tape Recorder Equipment Profile (May 1968)

It was erroneously stated in reviewing the Sony/Superscope Model 230 four-track stereo tape recorder that it had a signal-to-noise ratio lower than that claimed by the manufacturer, where, in fact, S/N was higher. This variance in measurement was due to using different references. Whereas AUDIO employed a reference point of 0 VU at 1% distortion, Sony/Superscope's specifications clearly note that measurements were taken at *peak level*, which would be at the 3% mark commonly used by many tape recorder manufacturers. This would indeed enable the machine to easily meet its signal-to-noise specifications since it increases the ratio by 6 to 8 dB.

Also worth noting is omission of mention of the machine's inclusion of a "scrape flutter filter." Usually found only in professional recording equipment, the "scrape flutter filter" is a special idler located between erase and record/playback heads. Its purpose is to eliminate tape-modulation distortion.

## BSR McDonald Model 600/M44-E Automatic Turntable

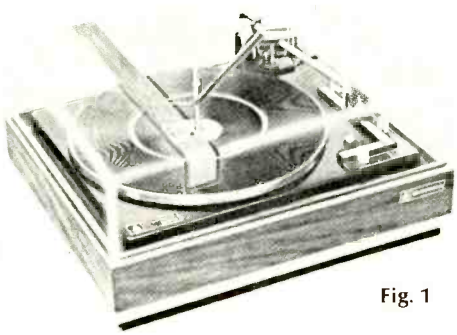

Fig. 1

**MANUFACTURER'S SPECIFICATIONS — Speeds: Four. Platter Diameter: 11". Wow (at 33⅓ rpm): 0.1% rms. Flutter (at 33⅓ rpm): 0.04% rms. Tracking Error: Two deg. max. Stylus-Force Range: 0-6 gms. Arm Resonance: 15 Hz. Price: $89.50, including Shure M44-E cartridge, WB-6 deluxe base and DC-3 deluxe dust cover.**

The BSR 600/M44-E is a complete record-playing unit. That is, it consists of an automatic turntable with a Shure Model M44-E elliptical-stylus stereo cartridge already installed, and a walnut-finish wood base topped by a plastic dust cover.

The Model 600 changer used here is the top unit in a line of moderately priced BSR McDonald record changers. The changer itself has a retail price of $74.50. In contrast, the whole package is available for only $15.00 more (if bought separately, the components and accessories would total $128.00.)

The turntable incorporates most of the features of higher-priced automatic turntables, though in some cases in a more simplified way. The changer unit features a low-mass tubular arm that comes with a sectional counterweight to balance against a wide range of cartridge weights. Both counterweights are resiliently mounted and are easily adjustable for see-saw balance before the tracking force is "dialed." The arm's ball-bearing pivot is angled parallel to the plane of the clip-in cartridge shell. There are 2 spindles: One short one for manual play and a long one for automatic play. Tracking force is set by dialing a number next to the arm pivot. The dial is calibrated in ⅓-gram increments. And it is pretty accurate once a good zero reference is established by sliding the rear counterweight to the proper place for balance. A most handy feature is the arm lock that automatically clamps the arm after shut-off and releases it during start. A muting switch shorts out the cartridge output during record changing and a "pop" filter eliminates motor on-off noises.

The cueing mechanism, which allows one to handle a lightweight tone arm manually without fear of dropping it on a record, operates with mechanical linkages. Therefore, the rate of tonearm descent is proportional to the speed with which the lever is thrown. While this method works well, it is not quite as effective as the pneumatic-type systems used in more expensive tonearm designs. However, this cueing device is an especially useful feature, meeting all but very precise tape recording needs.

Drive and changing mechanisms are rugged and simple. A four-pole induction motor has a stepped pulley mounted to its shaft, which drives a rubber interwheel. The wheel drives the inside of a sub-platter (7-in. diameter) that is riveted to the main 11-in. cast-aluminum platen. A novel speed-changing mechanism uses a nylon rack and pinion linkage to smoothly raise and lower the interwheel with the speed selector control, thereby lining it up with the different pulley steps. The entire mechanism was found to be jam-free and reliable in operation.

### Performance

The BSR 600/M44-E performed as follows: Rumble, including vertical and lateral components, was measured at −27 dB referred to 1.4 cm/sec at 100 Hz (or 3.54 cm/sec, 45 deg. velocity

Fig. 2—View of the BSR 600 with turntable platter removed.

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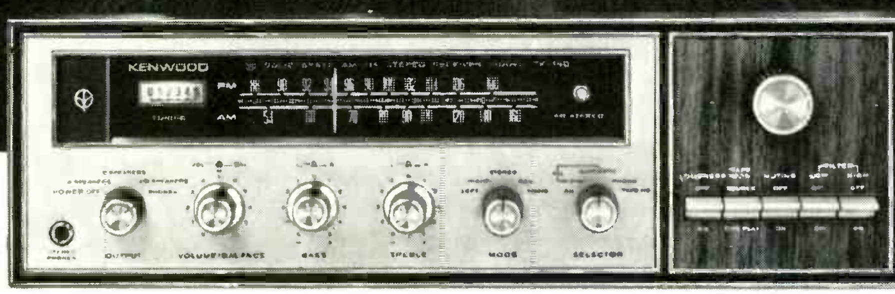


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at 1000 Hz), the standard NAB method for rumble measurement. With the vertical-rumble components cancelled by paralleling the cartridge outputs, the rumble was −31 dB. This is satisfactory for all but speaker systems that can exhibit great output in the very deep bass region. Wow was checked at 0.15% and flutter was about .04%. The speed at 33⅓ rpm was more than 1% fast, as well as running fast at the other speeds. Speed remained constant over a range of 85 to 135 volts, however, which is excellent. There was no arm resonance down to 20 Hz and the unit mounted to its base was not particularly sensitive to shock and vibration or acoustic feedback.

The Shure M44-E stereo cartridge which comes fitted to this BSR unit is the $34.50 member of the Shure family of high-performance ellipticals. It has the highest output voltage and, as an excellent performer at just under 3 grams of tracking force, appropriately complements the BSR 600 automatic turntable.

The sculptured walnut base of the BSR 600/M44-E package has an attractive metal sash around its girth, and the matching tinted plastic cover, with its distinctive walnut stripe, is effective in protecting the unit from dust and other hazards. In addition to presenting an attractive appearance, the BSR Model 600/M44-E succeeds admirably in filling the need for a modest-priced all-in-one record playing unit that incorporates many refinements normally found only in more expensive automatic turntables.

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## Jensen Model TF3B Speaker System

**MANUFACTURER'S SPECIFICATIONS – Frequency Response: 25 Hz-20 kHz. Crossover Frequencies: 2000 Hz, 10 kHz. Impedance: 8 ohms. Power Rating: 25 watts. Woofer Resonance: 30 Hz. Dimensions: 13½″ H x 23¾″ W x 11⅜″ D. Shipping Weight: 40 lbs. Price: Oiled Walnut, $122.00; unfinished, $109.00.**

The model TF-3B, a 4-speaker, 3-way bookshelf-size unit, is the least expensive one in Jensen Manufacturing's new series of five loudspeaker systems.

It contains four speakers—a 10-in. woofer, two 3½-in. midrange units, and a spherical-radiator tweeter. Housed in a dark walnut cabinet (an unfinished model is also available), it features dark olive-color cloth that matches walnut strips which divide the front into three sections. The outside two sections are covered with a gold metal grille, giving the unit a modern-style appearance. The metal grille also offers extra protection against accidental poking to the two mid-range speakers which are mounted behind one section. Two screw terminals (color-marked for polarity identification) are recessed in the rear of the enclosure, together with knurled shaft of a high-frequency level control.

The cabinet of the TF-3B is made of a plywood-flakeboard walnut veneer combination, ¾-in. thick. The backs of the mid-range and tweeter speakers are sealed with metal, an integral part of the speaker "basket" assembly. The woofer, which has a 1¼-lb. magnet, utilizes the enclosure's ducted port (a heavy cardboard tube, 7-in. long and about 3-in. in diameter) for bass loading; a rubber gasket ensures a tight seal of the enclosure rear. Crossovers are of the L-C (inductor-capacitor) type. The high-frequency-level potentiometer controls both mid-range and tweeter level, since mid-range crossover takes place at a high 2000 Hz.

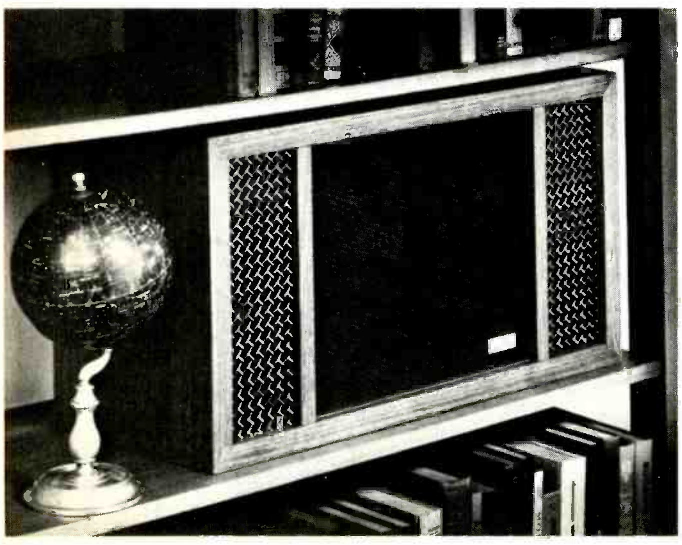
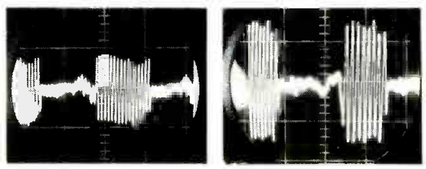


Fig. 1—Jensen Model TF3B bookshelf speaker system.

### Performance

With the HF level control turned up all the way, the measured frequency response of the TF-3B averaged about ±6 dB between 60 and 16,000 Hz. This is particularly fine. Response dropped off below 60 Hz and doubling could be induced below 50 Hz at high input levels. At low levels, we could measure output down to 30 Hz. There were no significant peaks or dips in the response. Tone bursts, shown in Fig. 2, back up the excellent transient response of the speakers, with no evidence of ringing anywhere. The high-frequency dispersion was good, as would be expected from the dome-type super tweeter. In a hard room, one might turn down the HF level control just a bit. Otherwise, full up is OK. Efficiency of the speaker system is low. Therefore, we recommend a 25-watt (rms) amplifier for use in a 12 by 18 ft. room. In a smaller room, a lower-power amplifier would suffice, of course.

Fig. 2—Tone bursts taken at 250 Hz and 10 kHz.



In listening tests, we found that the TF-3B had a full, warm sound. There was a slight tendency to be boomy at very high listening levels, but a speaker of this type should not be used to produce sound pressure levels that can burst ear drums. It offered its best sound in a small room, at medium and lower acoustic output levels. When listening to a stereo pair of TF3B's, we observed excellent stereo balance, with no wandering or peaking, which denotes close similarity in the two units.

The TF-3B, therefore, turns out to be an excellent performer in its class. It should serve well in complementing a medium-priced sound system. Persons who want a moderate-sized bookshelf speaker system with a little more sculptured face to go along with fine performance will find the TF-3B very appealing, indeed.

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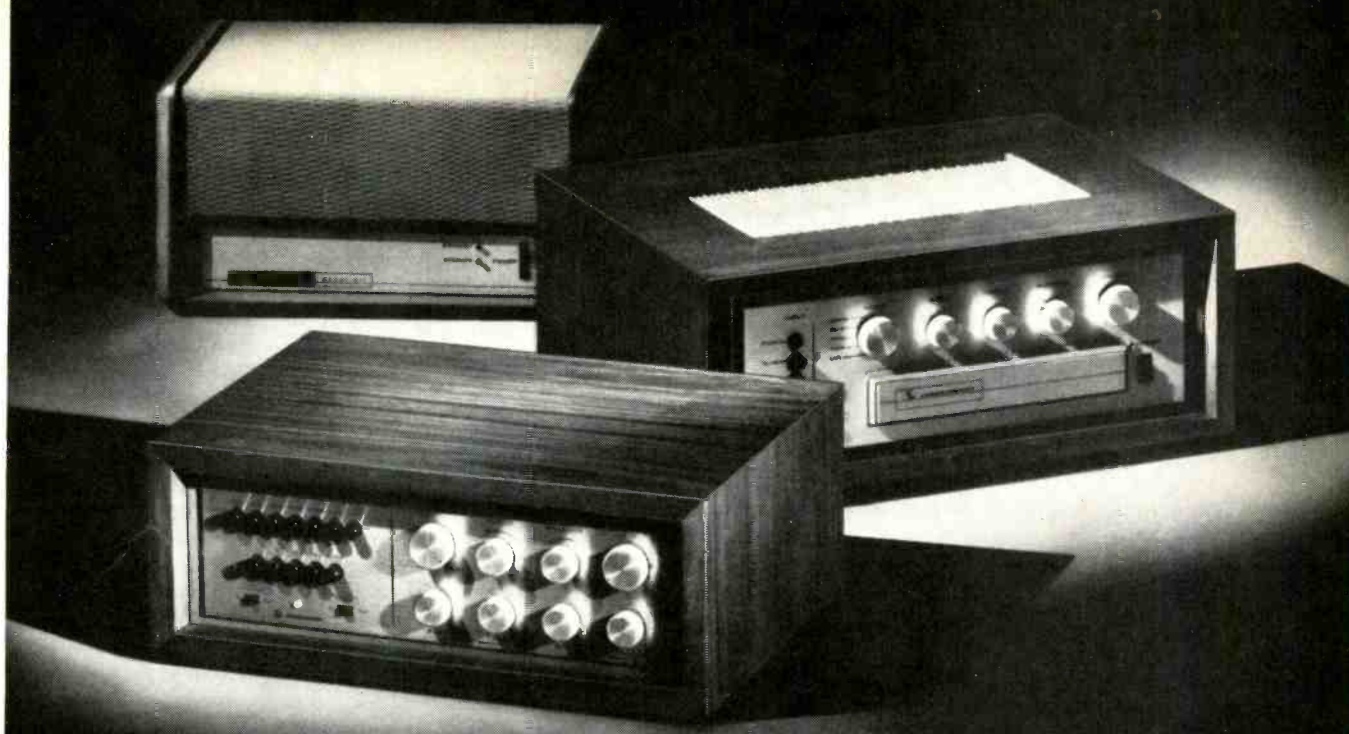

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# AUDIO MUSIC REVIEW



# Classical Record Reviews

EDWARD TATNALL CANBY

## Baroque Organ

**Music of Frescobaldi, Sweelinck, Bach.** Lawrence Moe, organist.

**Cambridge CRS 2513 stereo**

This is a first rate "Baroque" organ record, from the University of California, offering not only excellent hi-fi sound but two very different organs and music to match each. Plus an organist, who for once, is really musical in his playing, shapes each line with care, gets through to the meaning of every section, plays a superb *legato* and a dynamic *staccato* and registers the music with splendid color structural sense. *Cambridge Records* is well known for this kind of excellence. It is one of those rare, small companies that continue to exist on a very few releases, every one of real interest. Don't know how they do it.

Side 1 offers a lovely little "chamber organ," built in 1783 but using pipes fabricated even earlier. The appropriate music is by Frescobaldi and Sweelinck; Frescobaldi, one of the very earliest true organ composers, seldom has sounded so convincing and the lovely Sweelinck variation (Dutch) on a folk song, "Mein junges Leben hat ein End" are faultlessly set forth, with never a note unphrased or uncalculated yet with human warmth rare on the organ —a mechanism that must always, of course, be forcibly "humanized."

Side 2, on a larger modern Holtkamp organ in the high Baroque style, offers a Bach program, a prelude and fugue (A major) followed by the Christmas Canonic Variations on "Vom Himmel Hoch," a familiar chorale melody. Again, excellent phrasing and shaping of every detail and none of that arbitrary staccato playing that too many organists do by habit, after years of dull-toned and blurred church organs. (Only way to punch the melody through the murk—but not on *this* instrument.)

Note *RCA*'s better-known Weinrich recordings on a similar Holtkamp organ in New York—this man Moe is a far finer organist. Only RCA's big live acoustics are superior; this organ is in a somewhat lifeless small hall.

*Performance:* A — *Sound:* B+

## A Jazz-Classics Mix

**Charlie Byrd—Music of Villa-Lobos. (Preludes Nos. 1-5; Etudes Nos. 1, 5, 6, 8, 11).** Guitar solo.

**Columbia CS 9582 stereo** ($4.79)

I haven't kept up with the raft of previous Byrdish albums that are in the Columbia "popular" catalogue. All I know officially is that here is a famed jazzman playing "classical," all by himself, not even a lush orchestral backing.

Easy enough to pass judgment! It's excellent. Maybe even better than Columbia knows. Jazz or no jazz, this man is a first rate "classical" guitarist, technically very much on top but, more important, with an easy musicianship that makes the music sing with really effortless grace and naturalness. A thorough pleasure, absolutely unqualified.

Not that Villia-Lobos (Brazilian) is very heavy "classical." Far from it. Like so many Latins, he was prolific and eclectic, with no great worries about being either profound or difficult. The early Preludes (1929) are more or less modified Spanish-school, diomatic for the guitar and totally easy on the ear. The later Etudes on Side 2 (1940) are more complex, with some moderately biting dissonance; just pleasantly solid after the hors d'oeuvre of Side 1. A good album, period; and who cares what official category it's filed under. Maybe we ought to call it mood music. Darned good mood music!

*Performance:* A— *Sound:* B

**The Music of Ornette Coleman (Saints and Soldiers; Space Flight, Forms and Sounds).** With Phila. Woodwind Quintet, String Quartet of Ch. Symph. of Phila.

**RCA Victor LSC 2982 stereo** ($5.79)

This jazzman isn't playing classical —he's producing "classical" music (maybe) for a classical performers. And boy, is he trying hard.

Frankly, I found the stuff pretentious and self-conscious. By which I mean simply, that though it is full of extreme dissonance and makes a very important-sounding impact, what with the Philadelphia classical players and all, the music itself leaves me classically *very* cold.

I suppose this is partly because I am never happy when jazz people desert their relatively advantageous informality for the outward formal wear of the classical scene, white tie, black tails and all. That stuff is on the way out in "classical" music. Jazz is putting on the dog too late. In many ways, jazz is much stronger in its own style of music making. Reminds me of the "primitive" folk singers who want to give up their own genuine music in favor of show tunes and television.

Beyond the mere fact of this extremely dissonant and difficult classical idiom, played by such a forbidding array of ultra-ultra chamber players—a *string quartet,* mind you—is simply the music itself. I don't make *any* sense of it. Maybe you will, so better try. My ear is too popular-oriented, I guess.

*Performance:* ? *Sound:* B—

## Checkmate Farewell

**Wagner: Siegfried Idyll. Brahms: Serenade No. 2 in A, Op. 16.** South German Philharmonic Orch., Ristenpart.

**Checkmate C-76010 stereo**

Here, much delayed, is seemingly the last of the *Checkmate* releases from *Nonesuch-Elektra,* sealing what apparently was a miscalculation of considerable proportions—considering the enormous success of the original *None-*


**Most of the features of this $89.50 Dual were designed for more expensive Duals.**

You'd expect a big difference in performance between the $129.50 Dual, the $109.50 Dual, and the $89.50 Dual.

There isn't a big difference.

The higher-priced models have a few more features, but no more precision. Play all three through comparable hi-fi systems and we defy you to tell which is which, from the sound alone.

To achieve this similarity, Dual simply did what other manufacturers would get sued for doing. We copied the most expensive Dual.

We eliminated some things that weren't essential to the good performance. But we kept everything that was essential.

So, though we're about to describe the $89.50 Dual, the Model 1015, everything we say about it is also true of the more expensive Duals.

The 1015 has a low-mass, counterbalanced tonearm that tracks flawlessly with a force as low as half a gram. (Vertical bearing friction is .01 gram; horizontal bearing friction is .04 gram.)

The tonearm settings for balance, tracking force and anti-skating are continuously variable and dead-accurate.

The cue control is gentle and accurate, and works on both automatic and manual start. (Rate of descent is 0.5 cm/sec. The cueing is silicon-damped and piston-activated.)

The motor maintains constant speed within 0.1% even if line voltage varies from 80 to 135 volts.

Rumble, wow and flutter are inaudible, even at the highest volume levels.

If all we say about the $89.50 Dual is true, you may wonder why anyone would pay the extra $40 for the Dual 1019.

Perhaps there's something appealing about owning the very best there is.

**United Audio** Products, Inc.,
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Dual 1015

*such* label. So be it. I wrote the linear notes for these *Checkmates*, and I write them no more! Down goes the crew with the ship.

I wrote about the music but, by a technical quirk (no pressings yet). I never got to hear them. So this is my own first try at the release, liner notes or no. Interesting. Now just what does this disc (like the others in the series) have to offer? What is its problem, if any?

Technically it is a lovely job, an original recording (not merely licensed from other outfits, as are many *Nonesuch* releases), and it was done via the Dolby system, cut direct from the original non-mixed master tapes, recorded for stereo only. Fine idea, and still good as ever.

But musically there are problems. The big works of nineteenth century music—even these on the side—belong to the first-line orchestras and the powerhouse conductors, at least as we hear them in the U.S.A. We are in a sense spoiled. We have heard such music, if we know it well, in a long series of superbly tailored performances over the years, with all the advantages of the great-name leaders — from Toscanini onwards. Though we may not know it, we *can* tell the difference, when we hear merely a good, solid "small-town" performance. It's like a local "South Pacific" or "My Fair Lady"—excellent, but the original performance echoes through every note, in the mind.

Worse, sight unseen, we are equally spoiled; we'll buy the big names (especially in the low-cost reissues!) in preference to the small names, when it comes to the famous warhorses. That's the big-name problem.

Frankly, I found Ristenpart's Wagner very amateurish. He is not a specialist in this music; his best work is in the steady solidity of Baroque music, than which nothing could be more opposite than the "Idyll," with its sensuous, perfumed exoticism! It is nicely played here, on the surface. But there is no tension; it plods, it loses its way, its insides are poorly balanced, its marvelous changes of key and mood are perfunctory. The big conductors win by a mile.

Brahms, much more *gemütlich*, more friendly and German in this early Serenade, fares a good deal better. Nice contrasts, strings *vs.* winds. But even here, the tension is lacking, the music rather oddly paced (for those of us who have heard it elsewhere numerous times).

Good try, good idea; so-so realization.

*Performances:* C+, B — *Sound:* B+

## Violin Concertos

**Haydn: Violin Concertos (No. 1 in C; No. 3 in A, "Melk").** Nell Gotkovsky; Toulouse Ch. Orch., Auriacombe.

**Nonesuch H-71185 stereo**

In the summer of 1929 I was taken through the library of the fabulous monastery of Melk, on the Danube near Vienna—plastered cherubs and angels galore, a most un-librarian decor! Within a few feet of me lay the lost manuscript now known as the Melk concerto, discovered there almost twenty years later and recorded on one of the early LPs of the 1950s. It is a work of the first ten years of Haydn's composing life, that sprightly period now becoming so popular on discs, and will remind you of Mozart as much as of the later Haydn. Mozart was a child when it was composed. The other concerto on this record is from the same period, a bit earlier. Both make lovely and effortless listening, either foreground or background as you wish.

Miss Gotkovsky is an ideal Haydn violinist, superbly accurate in her pitch and rhythm and in her effortless double stops, entirely free of the flamboyant Romantic trickery that so many older violinists still employ, yet a persuasive performer of expressive melody. Excellent! Even her cadenzas (by herself) are believable, which is something. Good orchestral backing, too, if in a rather over-large, though pleasing, acoustical surround.

*Performance:* A— *Sound:* B+

**Roger Sessions: Violin Concerto (1935).** Paul Zukofsky; Orch. Philh. de la Radiodiffusion-TV Francaise, Gunther Schuller.

**Composers Recordings CRI 220 USD stereo**

Composers Recordings is the electronic arm of the contemporary American composing fraternity (classical division). The label covers a cross section of current and recent production in the area. Faithfully reflecting the musical scene, the CRI offerings are wildly varied, in every imaginable style and of all degrees of content and accessibility —just so the CRI pro judges feel the music is representative of their collective art. The choices *are* authoritative, definitely, and many are widely valuable for listening, like this one. Others, perhaps, are musician's music, for the profession. That is to be expected.

*Not* this one, though you may never have heard of Roger Sessions. (I once studied with him, and taught as his assistant.) It isn't popular music, to be sure, but if you like, say, Prokofieff, and can take Stravinsky and Bartók, you will find the big Sessions Concerto, now already more than thirty years old, an easy work to listen to in terms of idiom. It is no longer very modern (though it is dissonant), of an elegant effect, neither jazzy nor twelve-tone, nor folksky. Solid symphonic concert stuff, the best of its kind, and sympathetically performed by a knowledgeable young violinist and a powerful young conductor.

*Performance:* A *Sound:* B

## Harpsichordist

**Rameau. Pièces de Clavecin 1724, Nouvelles Pièces 1728).** Kenneth Gilbert, harpsichord.

**Pirouette JAS 19036 stereo**

This forthright young Canadian harpsichordist plays an interesting instrument; modern but with its "quills" and all the rest modeled directly on 17th-century Flemish construction. It sounds more brilliant and bell-like than most modern instruments, and the annotation wisely suggests that playback volume be turned a bit low. Sounds *much* better that way since the instrument isn't as loud as it seems with full groove modulation. (It would be good if this admonition were more widely used in recordings of small-voiced intruments.) Turn it down and it is lovely. Up and it becomes unnaturally strident.

Gilbert is technically a whirlwind of a finger man, and musical, though his playing is far removed from the dramatic Romanticism of such as old Landowska—who played Rameau too. In the young style, he plays rather metrically, with a minimum of hesitations and ritards. But in the long run, after two sides, one comes really to respect him. The music is alive and expressive.

Note a typical scrambled omelette of commercial attributions here—it's a *Pirouette record*, on the cover, and a *Janus*. on the label, an *"Everest* production" distributed by *Ambassador Record Corp.*, pressed for *Baroque Records* (that's what it says) by *RCA Victor* Custom Division. And it comes from Canada! What next?

*Performance:* B+ *Sound:* B+

# Light Listening

STUART TRIFF

**Leontyne Price:** "Right As the Rain" and eleven other songs. Leontyne Price, soprano; arranged and conducted by André Previn.
**RCA Victor LM/LSC-2983** ($5.79)

Dear Miss Price:

Just heard your new album of pop songs, so let me say right off that we all love you and think you're great! And you don't have to prove to us what a regular gal you are by singing songs like "My Melancholy Baby" With that superb vocal instrument of yours, it's sporting, but really unnecessary for you to resort to imitations, such as that camp rendition (in German, yet!) of "Falling In Love Again."

Admittedly, "Hello, Young Lovers," "Sleepin' Bee," and "Sunrise, Sunset" are all fine songs, but don't you think they've been somewhat over-exposed? To be quite honest, I feel that you're not entirely at home in many of these numbers, judging by the manner in which you alter your style and timbre from song to song, sometimes, within the framework of the same song.

I hope we can meet someday soon, over one of my special martinis (nobody makes 'em drier—no vermouth), and talk about some of the wonderful and unduly neglected things by Gershwin, Weill, Arlen, etc., that would make excellent material for your second album of light repertoire. By the way, I'd like to nominate that song by the Previns ("Where, I Wonder") as the "sleeper" of this collection . . . a haunting and beautiful ballad, beautifully sung! With all good wishes. S.T.

*Performance:* B *Sound:* B+

**New Year's Concert**—Music of Johann and Josef Strauss. Vienna Philharmonic Orch./Willi Boskovsky.
**London CS-6555** ($5.79)

This is the seventh collection of music by the Strauss Family recorded by the Vienna Philharmonic under its concertmaster, Willi Boskovsky. It is every bit as welcome as the previous half-dozen and long may the series continue. Included in this present survey are three orchestral excerpts from rarely-heard operettas by the Waltz King: the Czardas from the third act of "Ritter Pásmán," and the Overtures to "Cagliostro In Vienna" and "The Queen's Lace Handkerchief." Tunes from the latter work are contained in the waltz, "Roses From the South."

Another Johann Strauss item of special interest is the waltz, "Karnivalbotschafter," which I believe is here making its debut on long-playing discs.

Josef Strauss, a fine musician, overshadowed by his more celebrated brother, is represented by two waltzes ("Dynamiden" and "Village Swallows") and two polkas ("Dragon-Fly" and "Windmill") which exemplify his expert craftsmanship.

The performances of all these works leave nothing to be desired. The Vienna Philharmonic is one of the world's great orchestras. And it sounds it here! The conducting is both authoritative and affectionate. London's reproduction is first-rate. Stereo effects are discreet except in the "Ritter Pásmán" selection, where violin and cymbalom solos are distinctly defined left and right. All in all, a thoroughly enjoyable record. S.T.

*Performance:* A *Sound:* A

## New Sonicspectaculars!

### by STUART TRIFF

LONDON-DECCA CREATED quite a sensation among sound buffs several years ago with the establishment of its *Phase-4 Series,* designed primarily to exploit the potential of the then novel medium of stereo. The accent was on sound; musical content was a lesser consideration. To a certain degree this is still true, but aural advances made by other companies have now reduced the novelty value of *Phase-4*. Realizing this, London has wisely been paying more attention to substantial repertoire, and more ambitious recent releases have been leaning heavily on symphonic chestnuts, performed by major British orchestras under well-known conductors.

With the inauguration of its new *Deram* label, London has more or less entered into competition with itself in the sound sweepstakes. Six discs, devoted mainly to current pop material, comprise the initial release. Once again, as with early *Phase-4,* the sound is the thing. The music, though pleasant and expertly played, is inconsequential, and the performing artists are something less than celebrated. So, there is little to divert the attention from the Series' sonic delights . . . which are considerable.

Now, for a brief description of the "Deramic Sound System" and what it is, and what it is not (advance publicity releases, notwithstanding). The recording technique employs two adjoining studios (for differing ambience) with a variety of microphones, each with different characteristics—the various sections of the orchestra requiring a minimum of four mikes. Twenty-two or more magnetic tape tracks are used during the process, along with twelve reverb systems and twelve British-made signal-to-noise reduction systems. I was somewhat more amused than impressed over this procedure on learning that the *Deram* people insist on referring to it as not being "mechanically-contrived" (!).

In common with *Phase-4,* the reproduction bears little real resemblance to the way an orchestra actually sounds in a concert hall. It is, however, a fabulous *phonographic* achievement. The record surfaces are exceptionally quiet and clean; each instrument is reproduced with X-ray clarity. The stereo perspective is broad and full-bodied. Another plus factor is the excellent sound quality obtainable at low listening levels. This will be a boon to the enthusiast who wants to show off his rig to friends, without making their conversation compete with the volume control—and this is as it should be in music which is of the background-mood variety.

Good as these records are, I cannot indulge their makers in two bits of wishful thinking made for them. As stated, the sound has very effective depth and dimension, but it most certainly does not "wrap itself around you." The other claim: ". . . as effective when moving around as sitting in one place," is true just so long as you don't move any further than from one end of the sofa to the other. I also feel obliged to point out that the playing time of these discs, which are not budget-priced, averages approximately 16 minutes per side.

**Orchestral in the Night**—Gordon Franks Orchestra (You Only Live Twice; A Walk in the Black Forest; Love in the Open Air; Brazil, etc.). **Deram** (stereo only) SML-13701 ($4.79).

**Voices in the Night**—Peter Knight Orchestra & Chorus (If I Were a Rich Man; My Cup Runneth Over; Puppet on a String; A Whiter Shade of Pale, etc.). **Deram** (stereo only) SML-13702 ($4.79).

**Latin in the Night**—David Whitaker Orchestra (The Look of Love; Corcovado; I Will Wait for You; Summer Samba, etc.). **Deram** (stereo only) SML-13703 ($4.79).

**Brass in the Night**—Tony Asborne's Three Brass Buttons (All You Need Is Love; Thoroughly Modern Millie; Helados; Cornflake; Sunspot, etc.). **Deram** (stereo only) SML-13704 ($4.79).

**Piano in the Night**—The Pianos of Tony Osborne (Volare; Exodus; Sentimental Journey; Elmer's Tune, etc.). **Deram** (stereo only) SML-13705 ($4.79).

**Strings in the Night**—Gordon Franks Orchestra (Viva Maria; A Man and a Woman; Umbrellas of Cherbourg; The Young Girls of Rochefort, etc.). **Deram** (stereo only) SML-13706 ($4.79).

*Performance* (for all): B+
*Sound* (for all): A+

# Jazz, etc.

### BERTRAM STANLEIGH

**Hank Jones & Oliver Nelson:** Happenings
**Impulse Stereo A-9132** ($5.98)

The Oliver Nelson band, with Clark Terry featured on trumpet, provides some supple backgrounds as Hank Jones creates delicate filagree patterns on his new Baldwin electronic harpsichord.

*Performance:* A *Sound:* A

**Jerry Hahn Quintet:** Ara-Be-In
**Changes Stereo 7001**

First release of a new offshoot of the Arhoolie label, Changes Records will concentrate on contemporary music. Hahn's quintet, comprised of former members of the John Handy and Charles Lloyd groups, integrates Afro-Indian influences into expressive, well-paced performances Sound is clean with a balanced spread. Changes Records is at P. O. Box 9195, Berkeley, California 94719.

*Performance:*B+ *Sound:* A

**Phil Woods:** Greek Cooking
**Impulse Mono A-9143** ($5.98)

Saxophonist Phil Woods supplements a seven piece jazz combo with performers on the *bazoukie, dumbeg,* and *oud* to achieve a new tonal palette. Themes from *Zobra the Greek, A Taste of Honey, Antony & Cleopatra,* and *Samson & Delilah* all respond well to the treatment, and the Woods group show themselves as interesting innovators.

*Performance:* B *Sound:* A

**Wild Bill Davis:** Midnight to Dawn
**RCA Victor Stereo LSP-3799** ($4.79)

Recorded live at Grace's Little Belmont in Atlantic City, the locale of his recent collaboration with Johnny Hodges, this platter features the same tenor, guitar, and drums combo of the earlier disc and much the same uninterrupted background of audience chatter. The group swings brightly, and the sound is about what one can hope for under the circumstances. But are such circumstances necessary?

*Performance:* A *Sound:* C—

**Bill Evans at Town Hall**, Volume 1
**Verve Stereo V6-8683** ($5.98)

Wonderfully communcative performances by the outstanding keyboard impressionist in jazz. His solo, in memory of his father who died two weeks before the concert, is one of the most profound jazz statements on disc.

*Performance:* A *Sound:* A

## Poetry, Rock, etc.

**Tim Buckley**: Goodbye and Hello
**Elektra Stereo EKS-7318** ($5.79)

In his second album, Buckley displays a more poetic, atmospheric personality than appeared in the vigorous initial platter. Ten eloquent, anguish tinged poems are chanted to an exotic, unearthly accompaniment.

*Performance:* A *Sound:* B

**Malachi**: Holy Music
**Verve Stereo V6-5024** ($5.98)

More psychedelic music with an East Indian influence, this disc offers a variety of slow, clear plucked sounds that provide an adequate background for some other activity but fail to keep the attention of this listener for more than a few minutes.

*Performance:* ? *Sound:* A

**Big Jim Sullivan**: Sitar Beat
**Mercury Stereo SR-1137** ($4.98)

A rocking beat, well-spread stereo, and a pleasant balance of exotic sounds combine to make most of this disc among the most engaging of the current flock of pop-rock-sitar recordings. Only the few bands with oboe and string arrangements keep this from being a total delight.

*Performance:* A to B *Sound:* A

**Movement Soul**
**ESP Mono ESP 1056**

Selections from mass meetings, sermons, rallies, and demonstrations, together with a number of individual interviews. Recorded in 1963 and 1964 in Alabama, Mississippi, Georgia, and Washington, D. C., these recordings of the spirit and music of the freedom movement are a well-prepared document with a full text in an accompanying booklet. Sound is remarkably good for field recording.

*Sound:* B





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# Behind the Scenes

**BERT WHYTE**

**The State of Open-Reel Pre-recorded Tape**

FOURTEEN YEARS AGO, RCA Victor became the first *major* record company to issue a pre-recorded stereophonic tape. The work was Richard Strauss' "Also Sprach Zarathustra," conducted by Fritz Reiner with the Chicago Symphony Orchestra. The format was open reel two-track stereo at a speed of 7½ ips. For all practical purposes it can be said that the stereo tape industry began with that remarkable recording.

Among the audio buffs of those days, "Zarathustra" and subsequent stereo tape releases created a sensation. It became a very "in" thing to "go stereo." But the price of admission to this exclusive club was high. You may recall that the tapes were astronomically priced (for example, a Tchaikovsky 6th Symphony was $18.95). Then there was the matter of the stereo tape machine, plus an extra amplifier and speaker. Unquestionably, stereo tape was for the affluent at that time, but backed by strong promotional pushes from manufacturers, this medium flourished although the growth pattern was steady rather than spectacular.

By 1957, which can be considered the "high water mark" of two-track stereo tape, sales were reported to have passed 5-million dollars a year. In spite of such drawbacks as poor signal-to-noise ratio and the "sput-phut" of d.c. nodule noise, stereo tape was exciting and offered a unique listening experience *not obtainable from a disc.* Near the end of the two-track stereo tape era, better quality control of the duplicating process and improved tape oxides gave us some tape recordings of remarkably high quality. The venerable phonograph disc, often the target of pundits predicting its demise, showed its amazing market vitality by adapting to stereophonic reproduction in 1958, and came very near to sounding the death knell for stereophonic tape. As you know, stereo tape survived with the introduction of the 4-track format.

Stereo on disc and on tape has now co-existed for 10 years, yet I am continually astonished that so many people insist on the two mediums competing with each other on a "warlike" basis. This attitude is especially prevalent since the advent of stereo tape in such new formats as open reel at 3¾ ips, the 8-track cartridge and the cassette. The general feeling is that technological advances in stereo tape have out-distanced the stereo disc. This posture is patently rediculous, of course. One can make a formidable case for the merits of stereo on tape or on disc, and an equally strong case for their respective disadvantages.

For the last ten years, stereo tape has meant the familiar open-reel 4-track format operating at 7½ inches-per-second. It is generally conceded that this type of pre-recorded tape affords us the highest quality of stereophonic sound available to the mass market. Given a carefully duplicated open-reel tape made from a really good master recording, we can enjoy wide range 30 Hz to 15 kHz sound on a remarkably consistent basis.

The 7½-ips speed bestows the advantages of good transient response, and lessens the problems of wow and flutter and scrape flutter. To a somewhat lesser extent, the speed improves the signal-to-noise ratio. Dynamic range can be quite wide, and a common advantage shared by all tape formats is that there is no distortion of the type caused by the inner grooves of a disc.

Needless to say, the longevity of tape—the ability to play the program repeatedly with virtually no audible deterioration—is a plus factor of all pre-recorded tape over discs. Tape escapes some of the distortion-causing factors inherent in record-playing equipment, such as inner-groove distortion. And some tape machines can reverse a tape's direction automatically—equivalent to flipping over a disc automatically—which puts tape ahead on this score. As to the disadvantages of open-reel 7½-ips tapes, there are some that are trivial and some that are troublesome.

The statement that open-reel tapes are "hard to handle" and "difficult to thread" must be given some relevancy, even though the experienced tape enthusiast may sneer at such ineptitude on the part of the "general public." The factor of price has always been a thorny point with pre-recorded tape. The gulf between the price of a disc and the identical program on a 7½-ips stereotape is considerable, and must certainly be reckoned as one of the reasons for the increasing interest in open-reel 3¾-ips tapes. Even taking into consideration the discount practices prevalent in most large cities, many 3¾-ips pre-recorded tapes have achieved a price parity with the stereo disc.

From the strictly technical viewpoint, the open-reel, 4-track, 7½-ips stereo tape suffers from two maladies. Crosstalk is one of the bugaboos, defined as a transformer-coupling phenomenon between adjacent pairs of head stacks. On a 4-track tape, tracks one and three, and two and four, are recorded in opposite directions. When one pair of tracks is producing music at a pianissimo level and the juxtaposed other tracks are unleashing a mighty fortissimo passage of music, crosstalk is at its worst. Most of the crosstalk is in the lower frequencies, but mid-range and even high frequencies can be involved. The severity of crosstalk varies with the type of music and the degree of quality control which was exercised in duplicating the tape. The application of a separate bias supply to each track in recent years has resulted in less crosstalk and in lower levels of crosstalk. Nonetheless, crosstalk remains an all too frequent accompaniment to the recorded program, and is often cited by many professionals and advanced audio buffs as the reason for their preference for two-track stereo.

The other technical shortcoming of open-reel pre-recorded stereo tape and, unfortunately, of every pre-recorded tape format, is the matter of signal-to-noise ratio (or if you prefer, tape hiss). More than anything else, tape hiss has been and is today, pre-recorded tape's chief flaw. The amount of tape hiss one perceives is subject to many variables. The size of the room in which the listening takes place, the acoustic characteristics of that room, the response and efficiency of the loudspeakers, and the sound level at which one plays back his tape can have a profound effect on what the auditor hears as a degree of tape hiss (and for that matter, of crosstalk and any other noises and distortions).

If a person lives in one of today's typically thin-walled apartments, which perforce imposes a limit on his playback levels, and he listens in a small room highly damped with carpeting and draperies, utilizing small speakers with limited bass/treble response, he can report with honesty that he does not hear tape hiss nor crosstalk. Contrast this with the audio buff who lives in a house and plays his tapes in a large living room through a wide-range system at a relatively high room-filling level. We can go a step further here, for if this person favors the use of some of the fairly inefficient acoustic suspension-type speakers, he must turn up his gain control more than if he used highly efficient theatre-derived or horn-loaded loudspeakers. Quite obviously there is a difference in hiss level, but even with the high-efficiency system, he might find the amount of tape hiss somewhat objectionable.

Admittedly, all the variables involved complicate the job of the reviewer of pre-recorded tape. Nevertheless, the critic should establish some frame of reference as to the level of tape hiss, crosstalk and other extraneous noises. I have used a sound-level meter to take readings in a number of locations in my listening room, of peak levels in the loudest portions of the music as played at a given level. By averaging these readings, I can get a decibel reference point at which I can assess the degree of noise. By adjusting the playback level of various recordings to this reference point, the differentials in tape hiss, crosstalk, etc. among the competing brands of pre-recorded tape can be ascertained with reasonable accuracy.

## Transition

Acknowledging all the merits and drawbacks of open-reel 7½-ips pre-recorded stereotapes, what is the present status of this medium and what of its future? These are not easy questions to answer. The entire field of pre-recorded tape is in a transitional phase. The backers of the newer tape formats are all trying to gain ascendency and become the dominant medium. They are thinking in "mass-market" terms, and I respectfully submit that this approach is not valid in respect to open-reel 7½-ips stereotape. I have always maintained that this product is for a quality-conscious market, especially for the devotee of classical music. Ampex evidently agrees with me. They are now the biggest producers of 7½-ips tapes, and I was told that they are in this market to stay. They are very positive in their convictions that the 7½-ips tapes are the preferred medium for their classical catalog as well as for high-quality pop material. The only 3¾-ips tapes in the Ampex classical catalog are productions made from such labels as "Everyman" and "Parliament," which are from low-priced stereo discs.

As far as open-reel 7½-ips stereotapes from companies other than Ampex are concerned, the picture is very bleak. Most surprising is that stereo pioneer RCA Victor has completely abandoned 7½-ips tapes. All their new material, both classical and pop, will be issued at the 3¾-ips speed. Catalog items will continue to be furnished at 7½ ips. For some time now, Capitol/Angel has been issuing all of its tape product at the 3¾-ips speed, and from what I can determine there is no plan to revive 7½-ips tapes, even for the classical catalog. Presumably, Columbia has some interest in the 3¾-ips market, as they have issued a few tapes at this speed. What their intent is in regard to the slow-speed material or their future plans for 7½-ips tapes I don't know.

It would appear on an overall basis that, although there seems to be some movement away from 7½-ips pre-recorded tapes, the massive strength of the Ampex conglomerate of labels will ensure a good supply of this product. There also seems to be a general feeling that if the problems of tape hiss and crosstalk could be overcome, the 7½-ips tape would not only stage a "comeback," but would win many new converts for this medium. Fortunately, there has been some progress in this direction. A few months ago Ampex introduced 7½-ips classical tapes duplicated with their new process called "EX-Plus." By optimizing their mastering equalization curve, the use of specially built record and playback amplifiers and a new type of monitoring meter that can accurately indicate maximum recording level, this permits the elimination of the 6 dB "safety buffer" formerly necessary to avoid distortion. This means that at the same listening level you used with conventionally duplicated tapes, the *EX-Plus* tapes permit a 50 per cent reduction of your volume control setting and a corresponding reduction in noise.

Does it help? Most assuredly, but with variable results. This is not so much a shortcoming of the new process, as it is with the quality of the masters that are furnished. With a good, quiet first-generation master, the *EX-Plus* tapes are excellent, with the best signal-to-noise ratio I have ever encountered on commercially pre-recorded tapes. The company has informed me that all of their classical material will now be duplicated with this process. They added that certain selected pop recordings will be given the *EX-Plus* treatment in the near future. Unquestionably, this type of duplication is a valuable ally in the battle against tape noise, but it is not the total answer.

Happily, the *EX-Plus* process will soon be augmented by the highy efficacious *Dolby* system. Ampex is awaiting delivery of their Dolby A-301 and, when it is installed, we will have at hand the means for the production of exceptionally quiet tapes. It is fairly safe to say that most major record companies are now, or shortly will be equipped with the *Dolby* system. It appears likely that those companies which are members of the Ampex tape combine will furnish them with copies of their masters in the Dolby "compressed" mode, and Ampex would prepare their dubbing masters from the "expansion" signal output of their Dolby A-301. If the record companies involved had made their original recording with the *Dolby* system, this would be ideal. On the other hand, it still would be possible to produce very quiet tapes even if the original recording was non-Dolby, providing the company made Dolby copies of the *original master* or at the very least, the *working master,* since the vast majority of either original or working masters have a signal-to-noise ratio of at least 60 dB. That there are other possible noise-reduction applications inherent in the Dolby system goes without saying. At the moment, the *Dolby/EX-Plus* combination appears to be the most practical key to the revitalization of the 7½-ips pre-recorded tape medium.

As previously noted, the open-reel, 4-track, 3¾-ips stereotapes are appearing on the market in appreciable quantities. As far as I can determine, the advantages of this medium are two-fold. One is that they are the first variety of stereophonic tape able to compete price-wise on a nearly equal basis with the stereo disc. The other is that the slower speed affords great continuity, especially of value in operatic and certain other types of recordings.

As is true of 7½-ips tapes, there is a considerable variation in the quality of pre-recorded 3¾-ips tapes. Some productions are impossibly noisy. I have heard a few . . . very few . . . 3¾-ips classical tapes in which the tape hiss was low enough to be "tolerable." In common with 7½-ips "pop" tapes, the 3¾-ips pop productions have such a limited dynamic range and are recorded at such relatively high levels that tape hiss is fairly low. In general, the 3¾-ips classical productions have more tape hiss, crosstalk, etc. than their counterpart at the higher speed. The reason for this is that the recording equalization used for the slower speed makes the tapes more susceptible to overload, so the engineers keep the peak levels appreciably below the maximum recording level. Add to this fact that 3¾-ips playback equalization is tipped up a bit on the high end, this combination of factors causes more noise.

As far as classical tapes are concerned, the 3¾-ips productions have other limitations in addition to their noise problems. Compared to 7½-ips tapes, they have less brilliance and clarity. They simply are not as clean-sounding. This is not so much a matter of frequency response, as many of the tapes extend to 9 or 10 kHz. Rather it is the area of transient response, the "fuzziness" and blunting of percussives, the dulling of attacks, that erodes the quality of the slow-speed tapes. This is particularly apparent with piano sounds.

The gremlin behind all this is "scrape flutter." This longitudinal oscillation of the tape occurs around 3000-3500 Hz, and in original recordings or one-to-one dubs, is not normally of much concern. However, the 3¾-ips pre-recorded tapes are dubbed at an 8-to-1 speed ratio and, unhappily, this drops the scrape flutter frequency to 240-280 Hz, right around middle C (256 Hz) and coincidentally the center of the piano keyboard. At 280 Hz, scrape flutter, with its phase modulation and frequency sideband distortions, raises hob, dulling transients and thus the clarity of the recording. To those who might be a bit skeptical about this, I will say that it is necessary to have a high-quality, wide-range playback system to fully appreciate this problem. To those who own such systems, if you can compare a piano recording on 3¾-ips tape with the same recording on 7½-ips tape, stereo I'm sure your ears will confirm the difference.

As far as eight-track stereo cartridges and the two-track stereo cassettes are concerned, they share the technical problems which afflict open-reel tapes, especially the noise, and may surpass them in severity. In addition, there are special problems peculiar to these new tape formats.

Deficiencies in pre-recorded tape notwithstanding, the tapes do offer many redeeming attributes, as mentioned earlier. Further, a tape recorder —that is, a tape unit with recording facilities—opens up a bright, new world of sound and fun for users. Æ

## Pre-recorded Tapes

BERT WHYTE

**Leontyne Price: Prima Donna, Volumes One and Two. RCA Victor TR35018,** 4 tr.-3¾ ips stereo open reel ($11.90)

For fans of Leontyne Price, this is a bonanza. No less than 18 excerpts are presented from operas ranging from Purcell's "Dido and Aeneas" and Handel's "Atlanta" to Puccini's "Suor Angelica" and Barber's "Vanessa." As a demonstration of this artist's versatility, it is a veritable tour de force. Her power is astonishing and seemingly limitless. Only at the extreme bottom of her lower register does she have a little trouble in projecting the voice, as was noticeable in her recording of the "Final Scene from Salome" on stereo disc. Otherwise she summons those golden tones with consummate ease and the singing is always pristine pure and dead on pitch.

Throughout this tape the voice is very bright and articulate with fine projection. There is good vocal localization and a judicious left/right orchestral balance. Vocal/orchestral balance okay. The acoustics are spacious and definition is such to ensure a lot of presence. Dynamics are moderately wide. When Leontyne hits some of those "high hard ones," however, we get some print-through. Sorry to report that the hiss level is too high in Volume One, and crosstalk frequent, although fortunately at relatively low level. Volume Two is much the same technically, with somewhat less hiss and the addition of some extraneous low-frequency noise that sounds like it might be "room-rumble."

This is beautiful singing, though, and once again one must say that the advantages of a higher-speed recording is sorely missed. I have long been one of the most ardent boosters of magnetic tape, but at the risk of being drummed out of the "tape corps," let me say that I have this identical album on stereo disc and, forgetting the factors of wear and "scratchability," in its present unsullied state it speaks for Miss Price more quietly and more eloquently than this tape does. B.W.

### CAPSULE COMMENTS:

**Jacqueline Du Pre playing the Elgar and Delius Cello Concertos. Angel Y1S36490** open reel, 3¾ ips—Beautiful music, lovely performance and good sound. Worthwhile if you can check your copy for an annoying hum. I have two copies . . . both have hum . . . maybe you'll be luckier.

## STEREO INSTALLATION

(Continued from page 26)

4. *Big Band Bossa Nova*—Enoch Light (Command RS844SD)
5. *Concertos by Bach, Vivaldi, Handel*—Yehudi Menuhim and the Bath Festival Orchestra (Angel S36103)
6. *Le Sacre du Printemps*—Orchestre National de la R.T.F. (Nonesuch H-71093)
7. *Beethoven Symphony No. 5*—Otto Klemperer and the Philharmonia Orchestra (Angel S35843)

These recordings provided a variety of types of music from soft, intimate, classical guitar to crisp, modern jazz to full symphony orchestra; the system performed admirably under all these conditions.

It can be seen from the calculation of reverberation time in the accompanying section (even though both the calculated time and the computed optimum time are approximate values, subject to subjective interpretation) that the room is well suited to general listening. It is possible, however, that "crisp" music such as piano selections might demand a slightly more reverberant environment. Reference to Fig. 4 reveals that there are three small rugs and one larger carpet on the hardwood floor. The appropriate calculations show that removal of the three smaller rugs would increase the reverberation time to about 0.61 sec., while removal of all four rugs would increase it further to about 0.7 sec. Such transformations are easily made, should the requirement arise; however, the lower time (0.58 sec.) would appear to be more suited to a general-purpose living room.

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References:

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## GLOSSARY

Here are some of the terms used in Dr. Dolby's article on audio noise reduction, which starts on page 19 of this issue.

*Attack time*—The time elapsing from the initial signal which inaugurates a control until the control is fully in effect. Usually expressed in milliseconds (ms).

*Breathing*—The sound caused by variation in gain of an amplifier if the filtering is not optimum. (See Swishing.)

*Bridging input*—Describing the input circuit of an amplifier which is designed to be connected across a line of a specified impedance, but not terminating the line, nor appreciably affecting its loading.

*Compressor*—A variable-gain amplifier designed to reduce the dynamic range of a signal by limiting the difference between the loudest and the softest sounds.

*De-emphasis*—A controlled change in frequency response applied to a signal which has been pre-emphasized previously. In FM radio, for example, a pre-emphasis is applied to the signal to boost the high frequencies in the transmitter, and a complimentary de-emphasis is introduced in the receiver to result in a flat signal in which noise is reduced appreciably.

*Diode clipper*—A diode arranged in a circuit that cuts off — hence "clips" — peak excursions of signal voltages to prevent overloading of succeeding circuits.

*Expander* — A variable-gain amplifier arranged to increase the dynamic range of a signal. Opposite of compressor. Both functions are often combined in one unit.

*Filter*—A circuit element designed to eliminate a specific range of frequencies from a signal. Low-pass, passes low frequencies and eliminates those above the specified frequency; band-pass, eliminates frequencies above and below the selected band; high-pass, eliminates low frequencies and passes highs.

*Floating* — Describes a circuit which has neither terminal at ground potential, nor does it have a center-tap ground. Used when the device may be connected to a circuit which may have a ground at some other point. Usually employed to eliminate ground loops.

*Pre-emphasis*—See de-emphasis.

*Print-through* — Each signal recorded on a magnetic tape represents a series of permanent magnets, which may affect the coating on adjacent layers of the tape. This undesirable effect is called "print-through."

*Recovery time* — The time, expressed in milliseconds (ms) that a circuit requires to return to normal after a controlling signal is terminated. (See attack time.)

*Swishing* — Noise generated by varying the gain of an amplifier rapidly. So-called because it is descriptive of the sound, which is more pronounced than "breathing."

*Threshold*—Signal level at which a control action is inaugurated. Below this level, the amplifier is in a normal linear condition.

*Tone burst* — Short, regularly spaced groups of a limited number of cycles of a continuous tone used in testing amplifiers and loudspeakers. Usual bursts contain 4, 8, or 16 cycles of the testing frequency, and output is viewed on an oscilloscope.

### NOISE-REDUCTION SYSTEM

(Continued from page 22)

to-back combination of germanium and silicon diodes in parallel with the resistor. This configuration provides the desired non-linear integration properties. Finally, the smoothed control signal is returned to the compressor and is used to control the current in the compressor diodes by means of Q203 and Q204.

The four compressor output signals returning to the control module are combined by the precision resistors R301-R304 and are fed back to the amplifier module. After being amplified by the feedback amplifier, Q105 and Q106, the resultant noise-reduction signal is fed to a switch which determines whether the processor operates in the recording or playback mode. For recording, the noise-reduction signal joins the main signal additively between Q104 and Q107; during playback, the noise-reduction signal is combined subtractively at Q102-Q103.

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1. E. T. Canby, "Audio ETC.," *Audio Magazine*, March 1967 and April 1967.
2. R. M. Dolby, "An Audio Noise Reduction System," *J. Audio Eng. Soc. 15*. 383 (1967).


NEXT MONTH: Dr. Dolby discusses applications of the noise-reduction system.

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## ABZ's of FM

(Continued from page 32)

often applied to the r.f. stage to prevent overload of the stage when particularly strong signals are applied. This means that the first stage must have the capability of exhibiting varying gain with different bias settings. This was easily accomplished with "variable mu" vacuum tubes and is equally easy to accomplish with today's transistors, which depend on bias for varying figures of gain. Note that this feature is called Automatic Gain Control, rather than AVC (Automatic Volume Control), the term used in AM receivers. This is because changing the *amplitude* of the r.f. signal in an FM receiver does not alter the volume or "loudness" of the resultant output unless we are speaking of signals so weak that they do not cause full limiting in subsequent i.f. stages.

Other features of an FM r.f. stage which are not apparent from examining the schematic alone should be noted. Coils represented in the usual manner in the schematic become just four or five turns of heavy wire, with rather large spacing between turns, often using air as a dielectric. The variable capacitor sections themselves usually have just two or three plates in the rotor or stator, since we are dealing with total capacitances of just a few picofarads. Dress and layout of parts is much more critical than in AM r.f. stages, because even an inch or so of excess wire length implies a significant amount of additional inductance. Proper grounding is very important, too.

The techniques used to design and lay out r.f. sections of an FM receiver have evolved over a great many years. It is not the sort of thing a novice kit-builder should attempt to do from simple referral to a schematic. It is for this reason, incidentally, that most tuner kit manufacturers supply the front end in pre-assembled form, often even pre-aligned. To really appreciate the differences between a broadcast-band r.f. design and one intended for FM reception, you should examine the front-end construction of an FM tuner. Note the overall shielding (good FM tuners usually enclose the entire front end in a metal shield — to preclude excessive radiation from the local oscillator and to prevent accidental or intentional tampering with precisely-aligned coils, trimmers, etc.).

Next, we shall bridge the great gap from tubes to solid state and discuss considerations in transistorized r.f. amplifier design which forced the re-education of a whole population of FM design engineers. Æ

## ADVERTISING INDEX