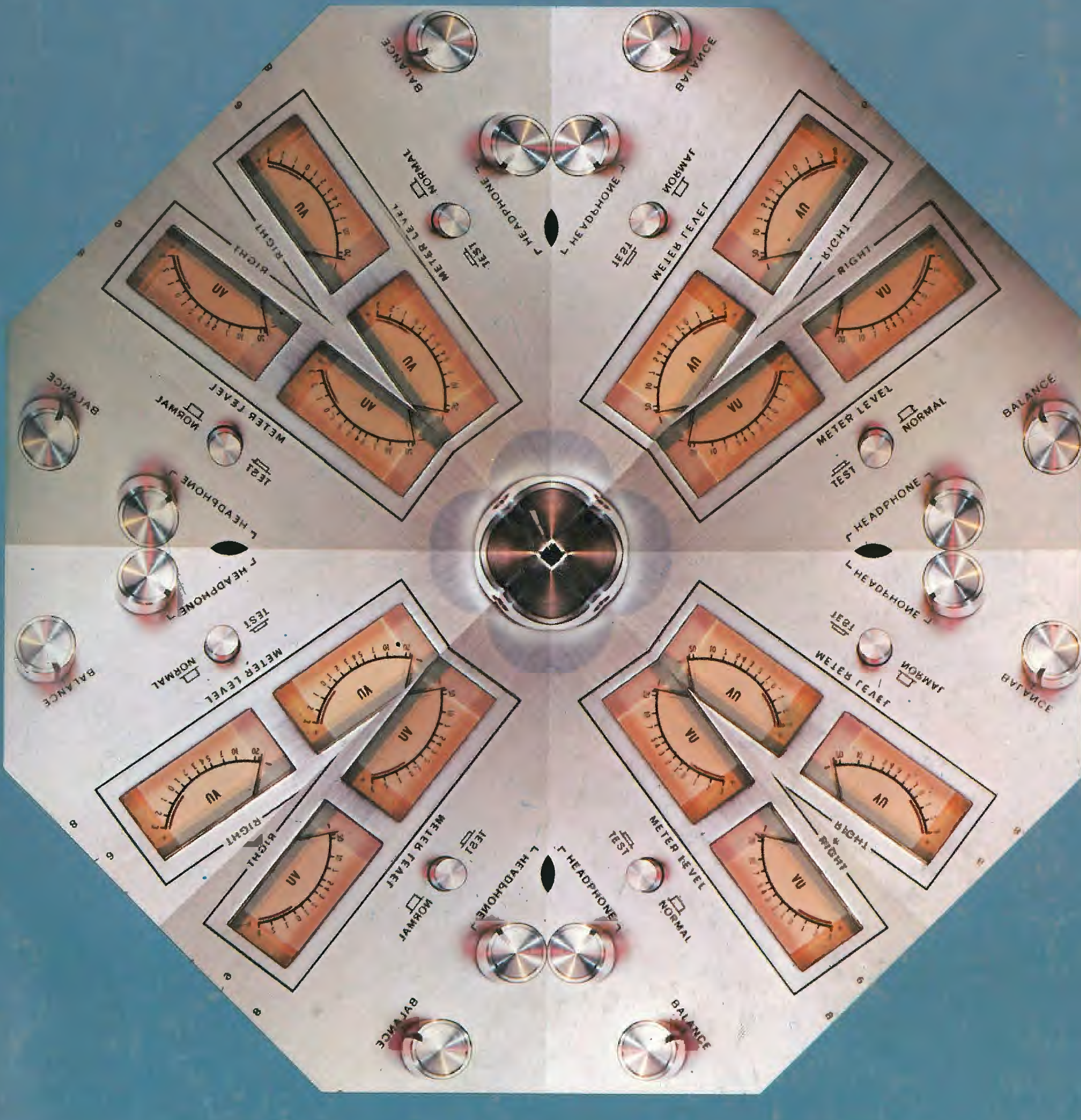
AUDIO

APRIL
1971
60c

The Authoritative Magazine About High Fidelity

SPECIAL
Transpet Devices

Guide to Amplifier Specifications
A Madsen-System Delay Tube
Canby Gets Another Head



POWER
and purpose are implicit in its every distinctive line...

Never before has there been a receiver like the 387. Power and purpose are implicit in its every distinctive line . . . from its bold new high-visibility dial face to the sweep of its comprehensive control panel. And just wait until you experience the 387's effortless performance! A new kind of receiver power is yours to command — instantaneous, undistorted, unmatched for flexibility and responsiveness.

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Decades of manufacturing experience and engineering skill have gone into the 387. But to really appreciate how its designers have totally rejected the ordinary, you must see it and hear it.

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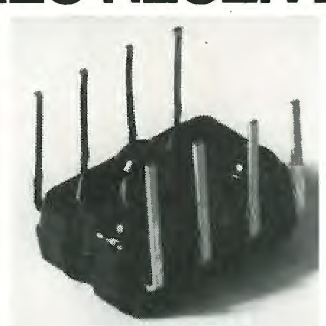
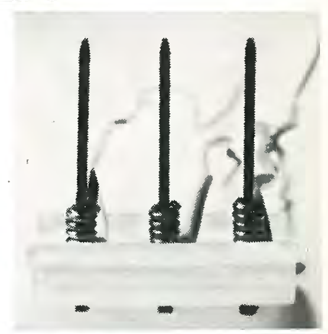


Ultra-reliable Integrated Circuits: Seven IC's are included in the 387 . . . totalling 91 transistors, 28 diodes, and 109 resistors.



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387 SPECIFICATIONS

AMPLIFIER SECTION: Total power (±1 dB) 270 Watts @ 4 Ohms; IHF music power, 220 Watts @ 4 Ohms; 140 Watts @ 8 Ohms; Continuous output, with one channel driven, 100/100 Watts @ 4 Ohms; 63/63 Watts @ 8 Ohms; Continuous output, with both channels driven, 85/85 Watts @ 4 Ohms; 55/55 Watts @ 8 Ohms; Harmonic distortion, 0.5% at rated output; IHF power bandwidth, 10 Hz — 38 kHz; Hum and noise, phone, —70 dB. TUNER SECTION: (FM); Usable sensitivity (IHF), 1.9 uV; Stereo separation, 40 dB; Capture ratio, 2.5 dB; Signal/Noise ratio, 65 dB; Cross modulation rejection, 80 dB; Selectivity, 42 dB. TUNER SECTION: (AM); Sensitivity (IHF), 4 uV @ 600 kHz; Selectivity (IHF), 32 dB.

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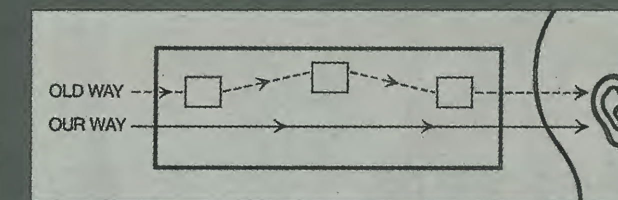

Now you can really snuggle up to Schumann. When you get next to our new stereo receiver, the SA-6500.

Because we cut down the distortion. By cutting out the input transformer, the output transformer and the output capacitor. So instead of putting your music through a whole electronic maze, we put it right through. Via direct coupling. With less than 0.5% distortion. And an amplifier frequency response of 10 to 100,000 Hz—1dB.

And because the signal doesn't get capacitored and transformered to death, you get something else. Full 200 watts of power (IHF) all the time.

The music is more than just close, it's sharp. Because we've got 1.8 μV sensitivity on FM from two 4-pole MOS FET's that can pull in your favorite station. So it sounds like it's being broadcast next door. Even if it's coming [illegible] the next state.

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AUDIO

Successor to RADIO, Est. 1917

APRIL 1971 — Vol. 54, No. 4

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EQUIPMENT REVIEWS

RECORD REVIEWS

AUDIO IN GENERAL

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Coming In May

Stereo Receiver Directory

Varactor Tuning
by Leonard Feldman

Equipment Reviews
Including:

AR-2ax speaker

Bogen BR-360 receiver

PLUS

Record and Tape Reviews
and
all the regular features

About the cover: This shows one of the very elaborate and versatile TA-2000 Sony preamplifiers. Photograph is kaleidoscopic—which word is derived from the Greek: kalos—beautiful, eidos—form, and scope—an instrument for looking. Any further remarks are unnecessary and could only get me into trouble. . . .

Audioclinic

JOSEPH GIOVANELLI

A Matter of Tracking Force

Q. I own a good record changer, but I am having trouble with its anti-skating adjustment. I use a cartridge fitted with an elyptical stylus, which tracks a 1.5 grams. I also have the antiskate control at this same 1.5 gram force. The problem is that the stylus tends to take long thin pieces of vinyl from some records and very little from others. Sometimes so much vinyl (not dust) collects on the stylus that I must stop the table to remove it. Shouldn't the anti-skating device prevent this? What effect will this "vinyl stripping" have on my records? What can I do about this?

I balanced the tonearm correctly and I bought a new stylus. The condition still continues. The antiskate adjustment is stable and will not move when set. Is this problem unusual?—Howard Wong, Woodside, N.Y.

A. The problem with your record changer is unusual. No vinyl should ever be removed by the stylus. I can only wish that your problem really is dust. It must be that you are tracking at a greater force than you realize. Do not depend on the gram scale on your changer. Obtain an independent force guage and try it. The anti-skating control does not have much to do with this situation. This device is designed to correct some distortion and to limit long-term record wear.

If you are unable to repair this changer, send it back to the manufacturer's service agency. You will ruin your records in short order if you continue using this equipment in its present operating condition.

Condenser and Dynamic Microphones

Q. What are the differences between condenser and dynamic microphones?—Arthur Darrow, Albany, N.Y.

The dynamic microphone is very similar to a loudspeaker. In fact, many intercoms use loudspeakers to double as microphones.

The dynamic microphone consists of a very low-mass diaphragm fitted with a coil. This coil is suspended in a magnetic circuit. This equipment is so constructed that sound pressure striking the diaphragm causes it to move in accordance with that pressure. This motion, in turn, results in the coil's moving. The turns of the coil thereby cut the flux lines of the field. The voltage is then passed along to the appropriate amplifying equipment for signal processing.

The capacitor, or condenser, microphone is much like an electrostatic speaker. (The dynamic and the electrostatic speaker and headphones were discussed in "Audioclinic," January, 1971.)

The capacitor mike has a very low-mass diaphragm, but with no coil attached. This is so arranged as to be one plate of a capacitor. Because the diaphragm can move as sound waves strike it, the distance between it and the other plate of this capacitor will vary in accordance with the sound pressure striking the diaphragm.

There are various schemes by which changes in capacitance can be translated into electrical output. One approach is to make the capacitance between the two plates in the microphone a portion of a tuned circuit. This circuit is then placed in a configuration such that it will oscillate. As sound strikes the diaphragm, the frequency of this oscillation will vary because of changes in the resonant frequency of the tuned circuit. The output of this oscillator will ultimately feed some kind of FM detector.

Another scheme is to place a large resistance in series with the capacitor and apply a voltage across this combination. As the diaphragm moves, the amount of charge on the capacitor will vary, causing a variation in voltage across the resistor. The variations of voltage across this resistor can be extracted and amplified. As was true of the dynamic microphone, this scheme results in a variation of voltage which is determined by the sound striking the diaphragm.

Minimum Response Requirements

Q. What does the frequency response of a music system need to be in order to faithfully reproduce music?—Sp/4 Don Niemczuk, APO San Francisco, Calif.

A. The minimum frequency response that is required for adequate sound reproduction is a highly debatable subject. Certainly we would like a system to be flat from 40 to 15,000 Hz. However, there are experts in this field who believe that an amplifier should by capable of reproducing flat to at least five times the highest audio frequency required. This would require a response flat to about 75 KHz.

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.

Transpet Devices

SPECIAL TO AUDIO MAGAZINE

Professor I. Lirpa, of the Bucharest Institute, has recently demonstrated a self-powered receiver, using his revolutionary transpets. Self-powered is not really the correct description, because the transpet devices get their power from cosmic rays. Figure 1 shows a schematic of a typical device. A is a polished rhodium disc that picks up cosmic radiation which is then converted to an electric potential by the ferrite-glass plate B. C is a sandwich layer forming the grid controlling the current flow to the anode D. Note the fins for heat dissipation. The space E, within the lead-lined glass bowl, is filled with krypton gas. Gains on the order of 95 dB are claimed. Larger devices are used for the output stages and Professor Lirpa recommends that the rhodium collector plates be mounted on the roof for best results. So far, power outputs of 10 watts at an impedance of 1 ohm have been obtained, but experiments are proceeding with graphite coated roofs which could generate power up to fifty times that figure.

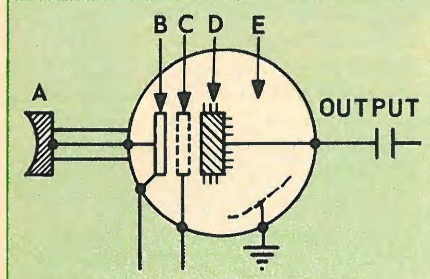


Fig. 1--Schematic of transpet.

Erratum

Open Reel Tape Recorder Directory
January, 1971, page 40
TEAC Corp. of America

Wow and flutter for models A-7010U and A-6010U should have read 0.08%. In addition, the company informs us, the "Special Features" comments for the A-7030U and A-7010U were transposed: Touch buttons, NAB hub adaptors, plug-in assembly, extra four-track play head, cue control, optional full remote control . . . apply to the A-7030U. The A-7010U is similar but is four-track with automatic phase-sensing reverse. To be strictly accurate, the A-1500 has a dual movement meter, rather than two meters as stated.

What's New in Audio

Erasette Model 200 B casette eraser

This cassette degaussing unit from Magnesonics Corp. is said to provide from 5 to 20 dB additional noise reduction depending on recorder and quality of cassette used. It is powered by four 6-volt "AA" batteries. A detachable cassette handle, provided with the unit, is used to pass the cassette over the eraser while an operating button is pressed. Signal removal is specified at -65 dB below 0 VU. Price, $15.95

Check No. 99 on Reader Service Card

LMF mid-range horn

This fiberglass horn from Community Light & Sound is said to be the world's most efficient mid-range speaker. The 12-in. driver accepts 150 watts rms input and delivers 134 dB over a 45 degree angle four feet from the mouth. Mouth diameter is 33 in., while total length is 40 in. A portable model breaks at the center and nests in a 23-in. length. Price, under $150.

Check No. 101 on Reader Service Card

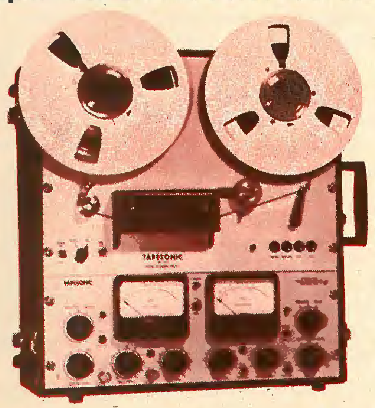
Mikado four-channel adapter

Mikado Electronics Corp. is not yet offering the four-channel adapter for conditional sales, as was mentioned in the January *What's New in Audio* section. Four prototype units are being placed in various locations in the San Francisco Bay area at no charge for testing purposes only, while the experimental transmissions are being carried out by radio station K101. Adapters will not be offered for sale until FCC approval is obtained for the system.

Ampex Dolbyized cassettes

Three additional Dolbyized cassette recordings have been released by Ampex Stereo Tapes, bringing to nine the total number of selections available on that label. The three are "Robert Merrill and The Prima Donnas," "Grieg: Peer Gynt Suite and Lyric Suite," and "Bizet Spectacular."

Tapesonic Series 70A recorders



These stereo recorders are all-silicon, solid-state designs with five modular plug-in units. Three speeds, 15, 7½, and 3¾ ips, three heads, A-B switch for each channel, electric push-button controls, three motors, automatic stop, and two mixing inputs are a few of the features. Frequency response specified at 15 ips is 35 to 26 kHz ± 2 dB; 7½ ips, 30 to 20 kHz ± 2 dB, and 3¾ ips, 30 to 10 kHz ± 3 dB. Signal-to-noise ratio is 56 dB at 15 ips, 53 dB at 7½ ips, and 50 dB at 3¾ ips. Available in ¼ and ½ track versions. Price, $675.

Check No. 102 on Reader Service Card

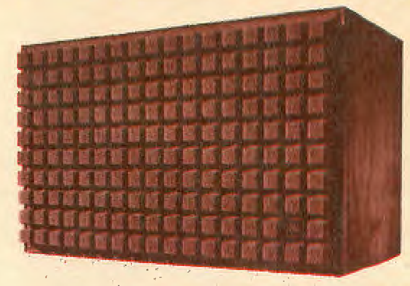
Microstatic speaker

This speaker system from Micro/Acoustics Corp. is designed to improve the radiation pattern and high frequency response of speakers limited in those areas. Four dynamic drivers are used to produce a 180 degree dispersion pattern. Response of the unit is specified at 3.5 to 22 kHz ± 2 dB with harmonic distortion less than 0.45% below 15 kHz. Price, $77.

Check No. 103 on Reader Service Card

JBL L100 Century speaker system

This home version of the JBL 4310 studio monitor is said to have an on-axis response of 40 to 15 kHz ± 3 dB. It is a three-way system, incorporating a long-



excursion 12-in. woofer with a free air cone resonance of 27 Hz, a four-in. mid-range unit operating from 2500 to 7500 Hz, and a direct radiator, crossing over at 7 kHz. Price, $264.00

Check No. 104 on Reader Service Card

Kenwood KT-5000 tuner, KT-5002 amplifier

Specifications for these stereo units are: Tuner, FM sensitivity (IHF), 1.7 μV; FM frequency response, 20 to 15 kHz +0, -2 dB; FM harmonic distortion, mono less than 0.6%, stereo, less than 0.9%; FM capture ratio (IHF), 2.5 dB. The amplifier is rated at 30 watts

rms per channel at 8 ohms from 20 to 20 kHz; with IM distortion, less than 0.3%; harmonic distortion, 0.5%, and frequency response, 20 to 50 kHz ± 1 dB. Prices, tuner, $179.95; amplifier, $219.95.

Check No. 105 on Reader Service Card

Triadex Muse computer synthesizer

This unit composes and plays music through a program established with four interval switches and four theme switches. Potential note combinations are said to be more than 14 trillion! It can be operated with a built-in speaker or an optional external amplifier-speaker. The Muse can be employed to teach scales, intervals, melodic construction and variation, imitation, etc., and can be used as input for a synthesizer, tape deck, amplifier, or head set. In a rhythm section it will play bass lines in a diatonic major scale. Price, $300.

Check No. 106 on Reader Service Card

Tape Guide

HERMAN BURSTEIN

Left-Channel Dropout

Q. The problem I have occurs only with certain tapes. With some tapes my two meters read the same output as they do input (switching from tape to source to tape, etc.), and they are balanced on both input and output. However, with other tapes, Channel A drops below Channel B, sometimes about 2-3 dB, and other times about 4-5 dB. Any suggestions you can offer would be very much appreciated.—John D. Moss, Hartselle, Alabama.

A. You may be experiencing what is sometimes called "left channel dropout," owing to imperfect contact between the tape and the heads. The middle portion of the tape tends to make better contact than does the upper or lower edge. Since the left channel is recorded on the upper or lower edge, this channel tends to produce less output owing to poorer contact. Such contact depends in part upon the smoothness and pliability of the tape. Inasmuch as these characteristics may vary with brand and kind of tape, this may account for the varying results you get. Stay with the tape which gives you best results.

Direction Changing

Q. I have a project in mind that requires the use of a tape recorder that plays and records in both directions. Can a capstan motor and/or magnetic control units tolerate reversing at one minute intervals, or will they overheat? I know that in the case of many industrial type motors, starting and reversing under load would cause them to get very hot.—Paul Palmer, Del Mar, California.

A. My understanding is that the reversing tape recorders are made so that they can undergo frequent reversing. I have seen no warnings to the effect that reversing frequently will be harmful. After all, keep in mind that when users shuttle between fast forward and fast reverse in order to find a spot on the tape, and when reversing motors are used, these motors undergo considerable exercise. In other words, tape recorder motors are generally designed to take such exercise. And I would imagine that the capstan motor is similarly designed in the case of a reversible machine.

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

Demagnetizer Failure

Q. I have some questions concerning head demagnetization. I was in the process of demagnetizing the head of my tape deck when the demagnetizer failed to operate because of a faulty electrical socket. Does this mean my head has become permanently magnetized? Has this affected my erase head? There is now a very noticeable hiss in the recordings I have made since the demagnetizer failed.

I recently purchased an Audiotex test tape and found during the frequency test a considerable difference from the test tone in the bass region. At 200, 500, 1,000, 7,500, and 10,000 Hz, response was below the test tone. Does this indicate a worn head? If a head replacement is necessary, would you recommend one?—Bob Gorskey, Wheaton, Illinois.

A. It is possible that sudden collapse of the magnetic field of the demagnetizer did magnetize your head. However, I don't think that such magnetization, if it took place, is permanent. Try repeated application of the head demagnetizer to see if the situation improves.

The results obtained with your test tape do not necessarily indicate a worn playback head. If only high frequency response were below the test tone, then a worn head may be indicated. But you also get below the test tone at middle and low frequencies. Hence it is possible that either the frequency response of your tape recorder or the frequency response of the test tape is at fault. The only way to find out is to use a test tape recognized to conform to NAB standards.

In replacing a head, it is usually advisable to use the same type head as originally provided, particularly if the machine has a good recommendation. This column cannot make specific recommendations.

8-Track Erase Head

Q. I have several 8-track cartridge tape players, and I am interested in converting one of them so that I may record my own cartridges. I have no difficulties so far as the introduction of recording facilities goes. My problem lies in the availability of an 8-track erase head. To date I have been unable to find out if such erase heads are available commercially. Do you know if such heads are purchasable? If so, from where?—Peter G. Russell, Hope Valley, S.D.

A. I suggest that you write to two places: (1) to the manufacturer of your cartridge player; (2) to The Nortronics Co., Inc., 8101 Tenth Avenue N., Minneapolis, Minnesota 55427. Nortronics makes a large variety of replacement heads, and possibly it can help you in the matter of an erase head.

Cassette and Cartridge Wear

Q. I wonder if you care to comment on advertising that claims that cartridge and cassette tapes are "indestructible," will last forever, as opposed to records which get scratched and worn out. In view of the limited emphasis placed on head care in most recorder manuals, I wonder if a tape played over several years without cleaning the heads and demagnetizing them will do almost as much damage as a bad needle?—Laird Brown, Dayton, Ohio.

A. I think that your comments about casettes, etc. are appropriate. Deterioration tends to set in unless heads are cleaned and demagnetized. On the other hand, the performance standards of such machines are often appreciably lower than those of reel-to-reel machines, so that it takes a good deal longer before deterioration is noticeable. By that time, something else may likely have gone wrong, so that the heads will receive due attention when the cassette is brought in for repair.

Meter and Speed Problems

Q. We have a couple of problems with a Roberts Model 90 purchased in 1960 and overhauled in 1963. This unit has performed well with no complaint until recently, when the following problems were encountered: (1) The tape speed has slowed down; constant, but slower. In 1963 the drive motor was replaced; however, I would hate to put in another motor at a cost of $36, even if this is the cause of the trouble. The entire unit, including motor, has been thoroughly cleaned and lubricated, and the motor seems to be running fine. The belt drive is working correctly to all appearances, and as far as can be determined everything is free and not bound in any way. (2) The VU meter is inoperative, although checking with an ohmmeter indicates there is continuity. Before the meter "died," it would operate intermittently; gently tapping the front of the meter case would cause it to work again for a while.—Milo G. Burston, Alpena, Mich.

A. I am not sufficiently familiar with the mechanical aspects of your particular tape machine to give you very much help. A few thoughts do occur to me with respect to your speed problem: Are you operating under low line voltage? Some motors slow down under such conditions. Another possibility is that the shaft of the replacement motor, or a pulley attached to it, has a different diameter than the original equipment. This happened to me once when I had a motor replaced. Finally, is it possible that the capstan has worn sufficiently to reduce tape speed?

In the case of your VU meter, the only thought I have is that the pointer is stuck, and with care it might be freed. I suggest that you write to the manufacturer of the tape machine or of the meter for assistance on this problem.

Volume Control

Q. I am considering a cassette tape recorder, Aiwa TR-1009. I plan to install it in a Zenith stereo model SFH2505T, with amplifier chassis 5G29. The only problem is: Would I be able to control the volume?—Michael J. Bitondo, Cliffside Park, N.J.

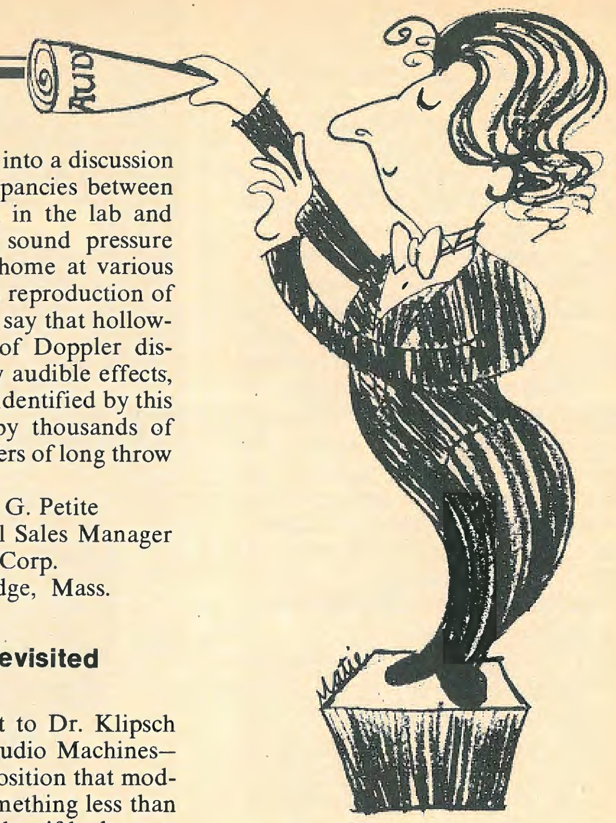
A. Every item of audio equipment I have ever encountered in the way of tape recorders and stereo sets includes a volume control. Therefore I do not see why you would have a problem.

Adding VU Meters

Q. I have two mixers that are used in a crude approximation of a console to feed two tape recorders. The VU meters on the recorders are difficult to observe due to the placement of the recorders in the room. To satisfactorily monitor each channel of each mixer, I would like to add VU meters to them. Could you suggest how to install these meters? Across the high impedance output? Use an additional preamp to drive them?—William B. Trigg, Smyrna, Tenn.

A. VU meters are of relatively low impedance. Hence they should not be used across a high impedance source. I suggest that you use an additional preamp, with low impedance output, to drive them. More specific information can be obtained very easily from the makers of VU meters.

Dear Editor,



Doppler Distortion

Dear Sir:

I read with interest Professor Greiner's comments on Doppler distortion in the December issue. According to his thesis, any long throw woofer whose response extends to 800 Hz or above should exhibit the hollowness in the voice range that he heard. However, there are numerous long throw woofers that cover such a range which do not sound hollow in the voice range; conversely, there are many speakers that theoretically have low Doppler distortion that do sound hollow—suggesting that the cause of the hollowness is something other than Doppler distortion.

Professor Greiner is correct in concluding that the problem occurs in the 500-1000 Hz area. Had his listening tests included a wide sampling of speakers he would have found that most of them sound hollow in the voice range, regardless of design. We have been able to ascribe this audible phenomenon to too much relative energy in this octave caused either by improper matching of octave to octave balance or more often by cone break-up products of the woofer introducing spurious energy in this octave. I believe he would find that this hollowness would still persist on test speakers if he used his electronic crossover to chop off the low end of the speaker's response, (that is, by using the crossover network as a high pass filter with a cut-off frequency of 100 or 200 Hz.), thereby proving that the cause of the hollowness cannot be ascribed to Doppler distortion.

If Doppler distortion isn't responsible for the hollowness, what then are its audible effects? The comments of AUDIO's editor on this question on page 28 of the August 1970 chronicle some of the long standing, wide-spread skepticism that indeed any exist! How can we explain the lack of audibility of the measured distortion Professor Greiner reported in his letter and others, most notably Paul Klipsch, have reported elsewhere; is the human ear that insensitive?

Perhaps the answer lies in the spectral content of music. Music is not flat! That is, it does not radiate equal power across the audio spectrum. It s acoustic power is typically 4 to 5 times less at frequencies below 100 Hz than at lower mid range frequencies. Therefore, the lab situation is strictly academic in nature and does not exist in real life.

One could go further into a discussion of the numerous discrepancies between the simulated situation in the lab and real life (such as the sound pressure levels required in the home at various frequencies for realistic reproduction of sound), but suffice it to say that hollowness is *not* the result of Doppler distortion. If there are any audible effects, they have not yet been identified by this writer nor I suspect by thousands of musically sensitive owners of long throw woofer systems.

Andrew G. Petite
National Sales Manager
Advent Corp.
Cambridge, Mass.

The Mud Factor Revisited

Dear Sir:

With all due respect to Dr. Klipsch and his Magnificent Audio Machines—and accepting the proposition that modulation distortion is something less than a good thing—one wonders if he has not inadvertently loaded his argument as well as his horns (The Mud Factor, October, 1970).

While a 100 decibel level at two feet might prove truly scintillating to a test rig, do we really listen at such outputs? Particularly at 42 Hz?

Without getting lost in the wilds of the Klipsch-Villchur Factor (KVF), and regardless of the intentions of learned discussions—as well as salesmen's clumsy demonstrations—the message seems to come through is that almost any sound reproducer will do better if the burden on it is kept light. Further, one's own listening preferences have to be considered—i.e., on the one hand, almost any sort of speaker will do for rock, or for that matter the well recorded but very palid background musics, but full symphonics seem to remain a problem.

All this leads me to wonder whether a collection of (probably small) individually boxed speakers, with amplifiers tailored to deliver various segments of the range to the different outlets, might not be better than further harangues about the relative merits—and sizes—of two-way and three-way packaged "systems." By extension, this could be carried to the point of idiocy, but from discussions in your publication, it would seem the required idea men, engineering, and equipment are on hand for a whole new shake, in the name of clarity.

Robert Cummins
Philadelphia, Pa.

Stokowski's Baton

Dear Sir:

I had to drop you a line after reading my January issue of AUDIO. Mr. Weingarten states on page 58 that the music from Fantasia is played by the "Philadelphia Orchestra under the baton of Leopold Stokowski."

I make no claims as a great musical authority, but it is well known that Mr. Stokowski does not use a baton.

William P. Fink
Westminster, Md.

Dolbyization for FM

Dear Sir:

Judging from recent articles in your magazine and others, it appears that the Dolby noise reduction system in the type "B" format greatly improves the tape medium for home use.

It should also be noted that Dolbyization could improve FM broadcasts to the same degree. With this in mind, I propose that the Type "B" Dolby unit be incorporated into FM recievers of the future for use both in FM reception and tape playback. Some cost savings could be passed on to the consumer by elimination of power-supply duplication, although an automatic FM switching detector would undoubtedly add to the cost to some degree.

Marshall J. Grimm
Detroit, Mich.

BEHIND THE SCENES

BERT WHYTE

I'VE JUST made an exciting discovery that I believe will have a profound effect on the immediate future of four-channel stereo. You will recall that last month I gave you some preliminary details of a procedure in which two- and three-channel stereo tapes can be processed for four-channel stereo playback. This was developed by Mr. John Eargle, Chief Engineer of Mercury Records. I noted that Mr. Eargle had given me a sample of one of his processed four-channel tapes, and that much to the amazement (and a little chagrin) of a purist like me, the four-channel sound was eminently satisfactory. Since then, Mr. Eargle has made a few changes and refined his processing techniques. A few days ago he gave me two reels of tape in which the program material incorporates these refinements.

I have listened to these tapes over and over, because I still can't believe what I am hearing. I am literally open-mouthed in astonishment. My brain says no, but my ears tell me I am unquestionably and incontestably hearing the best four-channel stereo I have ever encountered! As an added fillip, most of the selections I listened to are familiar items from the Mercury catalog, which I *positively know* were *never* recorded in four-channel stereo. Thus I have been enjoying such gems as Stravinsky's "Scherzo a la Russe" and "Fireworks," Prokofiev's "Romeo and Juliet," Kodaly's "Hary Janos Suite," Gershwin's "Strike Up The Band," and shades of the Late, Late Show, Leroy Anderson's "Syncopated Clock." All of the music I heard was either classical or pop material recorded in the classic mode. The goal of this four-channel stereo is the enhancement of acoustic perspective by means of ambient information, which is normally derived from the hall by recording two rear channels, but which in this case is the result of Mr. Eargle's special signal processing techniques. Thus Mr. Eargle's processing is applicable only to classical music or to pop music which would be recorded in a "ballroom" or concert hall configuration in which ambient information would be an integral part of the recording. The Eargle process would have no application in pop music of the equal intensity, ping-pong, surround type of four-channel stereo. Is this a drawback? Not at all, since in any four-channel playback medium, whether it be tape or disc or FM, pop and classical music are already mutually exclusive. It is important to remember that the product of the Eargle process is a four-channel stereo in-line tape, with the channels every bit as discrete as if the music had been originally recorded in the four-channel mode. To play back one of these tapes, you need a four-channel stereo tape machine along with the four amplifiers and loudspeakers. Once you have an Eargle process tape it can be utilized for any four-channel purpose. . .it can be encoded onto a disc via a matrix system such as the Electro-Voice/Len Feldman system (which now seems to be the matrix system that is gaining new adherents every day) and decoded in the home. In this case you would be restoring the four channels from the two channels on the disc through the decoder and having done so, the four-channel information on the original Eargle process tape would be presented properly through the four amplifiers and loudspeakers.

Why the Eargle process works and how it is done, I'll cover shortly. What I want to emphasize at this point is that the four-channel stereo qualities of an Eargle process tape are easily and demonstrably superior to most recordings originally recorded in the four-channel mode. I know this seems hard to believe. . .but take my word that it is true. You have read my comments and the comments of others about how critical the front and rear balance is in setting up four-channel stereo in your home. Too much rear channels and you hear discrete instruments behind you, which is of course totally unreal. Too much front channels and you lose the ambient information in the rear altogether. It is quite subtle, and for this reason classical four-channel stereo demonstrations at hi-fi shows and in stores have been dismal flops. The noise levels are too high; the absorption factor of many bodies doesn't help. Most people say they can't hear anything in the rear. . . then if you turn up the rear channels so they can hear, they huffily state that you don't hear violins (or trumpets, or tympani, etc.) behind you. With an Eargle process tape, front and rear balance is a very simple matter, a task that requires but a few minutes. This is because the delay signals on the tape are virtually "custom-tailored" to the music and there is more of it to work with. Haas, and in recent years, Madsen, have demonstrated the precedence effect in delay signals, wherein there is localization of direct information at the front pair of speakers, while reverberation is sensed as originating at all four speakers. While it is possible to overbalance in favor of the rear channels, it has to be almost a deliberate act. There is no "vagueness" in this kind of sound, no super-critical adjustments. And a big plus is that you are free from the rigid, small, circumscribed spot where you must be to listen to present four-channel classical recordings. With the Eargle process stereo you can move around your listening room, sit off to the side, and you still are aware of the big concert hall perspective. That is the immediately distinguishing characteristic of this sound. . .you are transported to the acoustics of the concert hall and the sound you hear is unbelievable for it's absolutely stunning presence. This kind of four-channel stereo will be easy to demonstrate in the hi-fi shops and even at shows. In fact, I will be at the IHF Palo Alto, Calif. show at the beginning of April and I will have these tapes with me, so readers there can have a chance to hear what I am raving about.

The important thing about these Eargle process four-channel tapes is that in one dramatic fell swoop, the problem of four-channel software shortage has been solved. You have read in these pages and elsewhere, that we have plenty of four-channel playback equipment already on the market, but that the sum total of four-channel tapes is perhaps 20 or so. Now suddenly, we are not wondering what music will next appear in four-channel stereo format, but literally. . *.what would we like to have!* The Mercury catalog alone has countless stereo classical recordings, suitable for the Eargle process. Beyond this are the thousands of classical stereo recordings in the vaults of all the other record companies, and Mr. Eargle has graciously said that he will share his techniques with interested parties in those companies. Mr. Eargle modestly claims that no extraordinary skills are required in his process, but I would think that in certain matters of equalization and balance, good music taste and value judgments would seem to require a certain finesse. In any case, the easy availability of truly high quality four-channel tapes will have tremendous impact on the four-channel scene. Many projects held in abeyance because of lack of playback material will be revived, and I will make an unqualified prediction that once a lover of classical music hears these Eargle process tapes, he is just going

(Continued on page 14)

to flip and won't be able to resist the blandishments of a four-channel stereo system.

Now let us get down to the "nitty-gritty" of how these four-channel stereo tapes are processed. We are assuming that we are starting with a three-channel stereo tape master, since this format probably constitutes the main bulk of classical recordings in the vaults of most record companies. First off, the three-channel tape is center split in the normal fashion and we now have a two-channel stereo submaster. The output of this two-channel tape is fed into an Ampex tape recorder specially set up to operate at 80 inches per second. This gives a value of time delay on the order of 25 milliseconds and the output is feeding direct information to the rear channels. At the same time a polarity reversal is introduced in one of the rear channels. Mr. Eargle explains, "The reason for this is simply to keep delayed center information (the level of which has been determined by the musical demands of the front channels) from being localized as a phantom image between the speakers. This would of course depend largely on the rear speaker placement. The reversed polarity insures that correlated information in the rear pair will not be clearly localized, while at the same time the uncorrelated left and right information will not be affected." He goes on to state that the one value of time delay used to create the effect of early reflections, is apparently accepted by the ear; where the musical interest is held up front, apparently one repetition before the onset of reverberation suffices to create the impression of a multiplicity of early reflections. Here again the Haas effect is in action as the psychoacoustical equivalence of time-delay and relative levels of otherwise identical signals originating at two loudspeakers. Localization tends towards the earlier speakers even when there are substantial level imbalances favoring the delayed speakers. Thus, if the 35 millisecond limit of sound image fusion is not exceeded, then the rear channels can deliver an acoustical energy level virtually equal to that of the front speakers without interfering with the localization of direct information at the front speakers. This is what makes the balancing of front and rear speakers for playback in the home relatively easy as compared to most present four-channel tapes which have too much time-delay and thus impose more critical balancing problems.

We have reached the point in the process where reverberation is to be added. For this the well-known EMT-140 reverberation devices are utilized. The latest models of the EMT-140 have *two* outputs representing so-called "stereo reverberation." Two EMT-140 units are used. The purpose of what would appear to be an "extra" reverberation device, is to feed reverberation of a slightly different "color" into the front speakers, if the original recording was on the dry side, or in case the reprocessing engineer desired more flexibility in simulating a convincing reverberant field over four loudspeakers. Signals are combined and fed into the reverb devices monophonically. The outputs of each EMT-140 is essentially randomly correlated. Since each EMT-140 has two outputs (the stereo reverb), reverb can be fed to all four channels. If desired, the onset of reverberation can be delayed an additional 30 milli-

(Continued on page 44)

Editor's Review

THE WASHINGTON Hi Fi Show is over, and like the Westbury and Newton affairs it was highly successful. Attendance figures were well over 28,000 which compares more than favorably with the 22,000 for the last show held in the Nation's Capital back in 1969. Many of the rooms (at the Hotel Washington) were on the small side and acoustically left a lot to be desired. Even so, some very good sound was heard but those "barbed wire strings," and highly colored bass was also in evidence.

As in most shows of this kind, the milling crowds plus the high ambient noise level made it almost impossible to make a reasonable judgement of any equipment, but there was no question about the tremendous interest shown—especially in the various 4-channel demonstrations. Headphones were another attraction and so Koss, Stanton and Superex had three of the busiest (but quietest) rooms.

According to a Press leaflet. . ."Musically the Show will move more than it has in the past in the direction of progressive rock. Although classical records and tapes will also be used as sources, this repertory now accounts for only five per cent of the market." Very true, but I believe that people will be listening to Beethoven, Mozart and Bach long after rock is dead and forgotten. It is often said that music reflects the culture, the spirit of the age but the popularity of rock is as much a tribute to the enterprise of commercial interests with their high-powered, gimmick-conscious publicity boys as anything else. This does not mean that **all** rock is bad, after all, we must remember that many of our greatest symphonies were written for the money—although the rich patrons of the day were certainly more altruistic than our large corporations: However, it must be said that a great deal of rock music—hard, acid, progressive, or what-have-you—is moronic, discordant, and totally lacking in any inspirational values. Fortunately, the scene is changing and there has already been a revulsion against drug-glorifying lyrics and the bizarre groups who advocate "mind-blowing" self-destruction. Musically, themes from folksongs, jazz, and the classics are now heard more often above the beat, and there is also an attempt to find some semblance of spiritual values. Experiments are being made with all kinds of electronic music and, to quote Jon Landau*, a well-known critic, "There is a new audience that is going to require and demand a music for the 70's. No one yet knows. . . what the musicians will give them. But one thing is certain: the audience *will* have music."

Another point: Today's youth may be uninformed about classical music, they may be exploited by commercial interests, but they *are* interested in hi fi. More than that, they are extremely knowledgeable about the technical aspects—as any exhibitor at the recent shows can readily confirm.

* * *

The next show is the IHF sponsored event at Palo Alto. The place selected is the Cabana Hyatt House Motel and the dates are March 29 to April 5. This show is followed by the 40th Convention of the AES which will be held at the Los Angeles Hilton Hotel from April 27 to the 30th. No less than 15 technical sessions are scheduled with 75 papers.

* * *

The article on amplifiers (page 32) covers a great deal of ground, but even so, many sections such as those dealing with cross-over distortion, noise-level, and transient response could be amplified (ouch!) many times and still leave a lot unsaid. And so further articles on various aspects of amplifier design and performance will appear in future issues. How about construction projects? Well, we are working on those too, and details of a small IC amplifier will be printed in the May issue.

* * *

The Los Angeles Art Center of Design has announced that Joseph Tushinsky has joined the Executive Board. The college is widely recognized as one of the foremost schools of design in the world. Joseph Tushinsky, who is of course, the president of Sony-Superscope has a little-known musical background. At one time, he was a trumpet player in the NBC Symphony Orchestra under Toscanini, later becoming the producer and conductor of the Cargenie Hall Light Opera. After serving with the Army, he went to Hollywood and wrote the famous screen play "My Wild Irish Rose." Music is still his chief interest, and he has collected more than 5000 music rolls for the *vorsetzer* mechanical piano player.

* * *

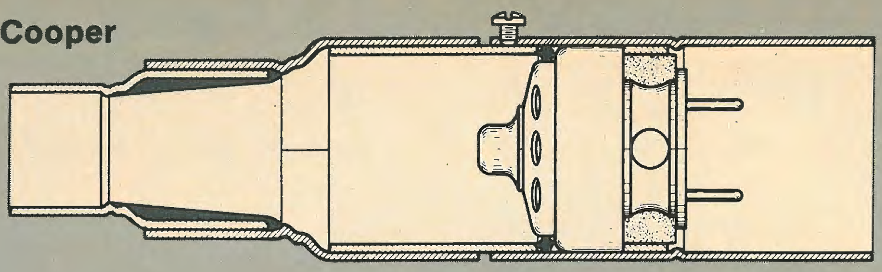
Humor in advertising

From a leaflet describing the Isotone loudspeaker". . .this new sound is for you who listen with eyes closed but with the inner ear keenly wide open in complete relaxation. . . . .listen-the-sound, it glows graciously even as it grumbles:" Grumbles? Maybe it needs a baffle. . . .

G.W.T.

*Newsweek, January 4 1971

Duane H. Cooper



Construction of a Madsen-System Delay Tube

This wide-range, time-delay unit adds ambience to a stereo system

AMBIANCE information already present in ordinary recordings may be extracted and convincingly presented via side loudspeakers by Madsen's method [1, 2] using a simple delay, as in Fig. 1. The method invokes the effects demonstrated by Damaske [3] that

(1) pairs of sources emitting incoherently-related sounds from widely-spaced directions fail to produce localizeable images, and that

(2), if one of these sources S_1 be in front, then the second source S_2 must be as strong as S_1 to avoid being masked when located in either the front or back, but that the masking capability of S_1 is 23 dB weaker if S_2 be placed to the side of the observer. Here, incoherently related sounds are differing superpositions of multiply-delayed replicas of the same sounds. In a reverberant enclosure, for example, the reverberant components observed at two widely-spaced locations would be incoherently-related replicas of the direct sound exciting the enclosure.

The simple delay used by Madsen exploits the precedence effect demonstrated by Haas [4] that, if S_2 be a simply-delayed replica of S_1, for delays in the range of 2 ms to 20 ms measured at the observer's location, then only the first sound will be heard (unless the second be very much the louder) as localized at S_1, though augmented in loudness because of S_2. Thus, although direct sounds are also presented from the side, the frontal image is not disturbed. Moreover, the "steering" of the side-presented direct sound around to the front unmasks the ambiance components, which are nonsteerable (Damaske effects), so that these are heard with the same spatial qualities, as near as the ears can tell, as in the original recording site.

The amount of delay used is the airborne propagation time from front to side, so that if the listener is more than two feet (about 2 ms) from the side loudspeaker, and the side loudspeaker is not too loud, then the requirements for the Haas effect are met, and the side loudspeakers do not appear to be "on," so far as the direct sound is concerned. (In small rooms, however, it may be advantageous to let the delay exceed that propagation time by as much as 5 ms.) Because the delay is so short, and thus not characteristic of the delays in the original recording enclosure, and also not a multiple delay characteristic of reverberation, it is clear that the perceived ambiance is not synthetic, except to the extent that the recorded ambiance had been synthetic. Very dry recordings still sound very dry.

Because the delay is so short (about 14 ms for a typical speaker arrangement), it is reasonable to consider using an acoustic wave-propagation tube to produce the delay. The problems associated with such tubes—frequency dependent losses and physical bulk—tend to be the more easily solved if long delays, e.g., 100 ms or more, are not required. A description of a bulky long-tube design has been given [5] in which the equalization problem was particularly severe, not only because the tube was long, but also because the application was reverberation simulation, involving multiple traversals of the tube excited by feedback. For single traversals, the equalization requirements need not be so stringent.

The design approach to be followed consists in the selection of standard low-cost components for transducers, tubing, and fittings, and to accept the design limitations thus imposed. The goal is to produce a satisfactory design that may be easily fulfilled without severely taxing home-workshop resources. This overall design is described in the next section, followed by sections on design considerations, the equalization techniques used, and points to be observed in placing the tube in service.

Overall Design and Construction

A schematic diagram of the acoustical tube arrangement is shown in Fig. 2. The numbers R-50 and R-70 are part numbers for microphone replacement cartridges manufactured by Shure Brothers, Inc. The driver element is the R-50 cartridge, a robust, low-cost unit of 1⅛ inch diameter that can dissipate a few hundred milliwatts without damage and can produce clean sounds with several tens of milliwatts of excitation. It is required to deliver acoustic power in the hundred-microwatt range. Its impedance is 200 ohms, almost entirely resistive at 1 kHz. The R-70 cartridge is used as a pickup microphone; it is a somewhat more costly unit having a diameter of ¾ inch and an impedance of 200 ohms.

The wave propagation tube is a 16-foot length of soft copper tubing in the half-inch size. Its actual OD (outside diameter) is ⅝ inch and its ID is about 9/16 inch. A size standard that is in common use quotes the size as a nominal ID (inside diameter) such that, in the heaviest weight, (1/16-inch wall) that ID would be the true value, and the OD would always be larger than the nominal size by ⅛ inch. In fact, even for a wall thickness of 1/32 inch, the standard OD is always larger than the nominal size by ⅛ inch. This tubing is supplied in 50-foot coils from which two acoustic delay tubes may be made.

For cutting the tubing, the smallest-size cutter was purchased, so that the cuts could be made with minimum deformation of the nearly uniform curvature of the manufactured coils. If substantial straightening and rebending of the coil were undertaken, an unworkmanlike-appearing coil would be the result because of nonuniform work hardening of the copper. A flat spiral coil with a close fit between turns is desired to help with later procedures needed to damp vibrations in the copper material. The as-manufactured coil has a radius of about one foot, and this is satisfactory, since tighter coils cannot be made anyway within the scope of home-workshop techniques.

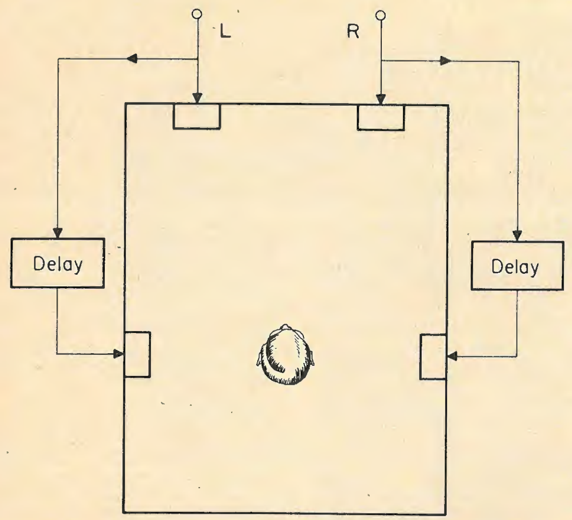


Fig. 1—The Madsen system of ambiance extraction. The side speakers are driven through a delay using the Haas effects to prevent them from affecting the direct-sound localization. They are placed to use the Damaske effects for a maximum presentation of non-localizeable ambiance.

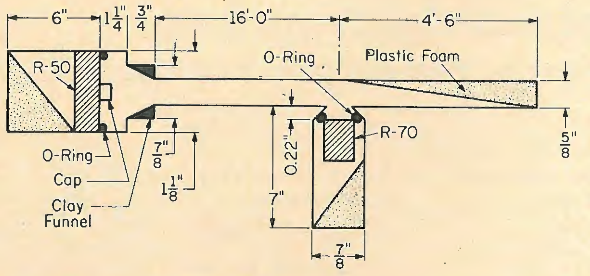


Fig. 2.—Acoustical schematic diagram of wave-propagation tube. The tube driver is the Shure R-50 magnetic-microphone cartridge, while the pickup is the Shure R-70 cartridge. Strippled areas show wedges of polyurethane foam for acoustical absorption. O-ring gaskets sealing the transducers to equalizing cavities are shown. Tubing diameters are OD.

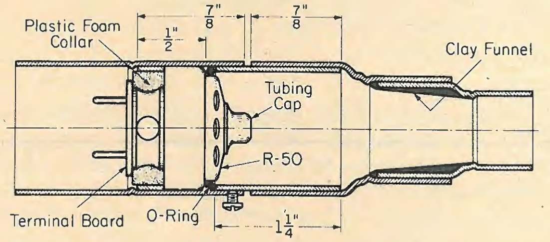


Fig. 3—Driver mounting. The R-50 driver is shown captive between an O-ring gasket in front and a quarter-inch collar made of one-inch tubing in back. The collar is retained by the internal ridge of the coupling and holds the damping material in a compressed state. A quarter-inch flare-fitting tubing cap is cemented to the front of the R-50. The front cavity is made of one-inch tubing inserted in the one-to-three-quarter coupling. A clay funnel is molded in the three-quarter-to-one-half coupling.

Steel electrical conduit in the ¾ inch size was used to make the pickup-microphone housing. Steel was used there, even though conduit size standards are incompatible with copper-tubing size standards, because the microphone housing contained the 10:1 step-up transformer (impedance ration 100:1), Shure part No. 51A34, and the magnetic shielding of the steel was desired. For a long run of microphone cable (more than three to six feet), a separately cased and shielded microphone transformer would be obtained. This is an Electro-Voice 502B, Shure A-95 series, or equivalent.

To adapt the conduit to copper-tubing standards, a ½ inch length of ¾ inch rigid copper tubing was cemented with epoxy to one end, and a ¾ inch copper coupling (solder-type), sawed in half, to the other. Since these are butt joints, it was necessary, to obtain sufficient strength, to cure the epoxy cement at an elevated temperature, and to make sure that the mating surfaces were clean (using paint solvent or "liquid sandpaper") and slightly roughened with sandpaper. Ordinary household epoxy may be cured in a cool oven (125°F. to 150°F.) for about 30 minutes. For extra strength, the joint to the coupling was subsequently packed with an outside layer of epoxy putty to a depth of ⅛ inch. So adapted, the microphone housing will accept a ¾ inch solder-type cap into which a connector is fitted, while the other end will fit over ¾ inch copper tubing.

The driver housing consists of a 1-inch copper cap, a 6-inch length of 1-inch rigid copper tubing (1⅛-inch OD), a 1-inch coupling, a 1¼ inch length of 1-inch rigid copper tubing, a 1-inch to ¾ inch adapter (C × C type), and a ¾-inch to ½-inch adapter (F × C type). All of these fittings are of the solder type, although no joints were to be made with solder. (Permanent joints were to be made ultimately with epoxy, for which a room-temperature curing is adequate for lap-type joints.) The "C" end of a fitting will accept an internal fit of the tubing, while the "F" end of a fitting will make an internal fit into the "C" end of another fitting.

The arrangement for coupling the driver to the tube is shown in Fig. 3. The ported back framing of the driver is wrapped with a single turn of ⅜ × ⅜-inch polyurethane-foam gasketing (3½ inch long) held in place with a ¼-inch collar made of 1-inch tubing. The foam plastic is ⅜ × ½-inch gasketing material (Macklanburg-Duncan Co.) that had been trimmed to ⅜-inch width; the paper backing was left in place to facilitate fitting the collar into place. Inside the 1-inch coupling, the front side of the driver is sealed to the 1¼ inch length of tubing by means of an O-ring gasket of 1⅛-inch OD and 1/16-inch thickness. Actually, a 1 3/16-inch OD ring was used, and a 3/16-inch-long section was cut out—just enough to place the butt joint in the ring under compression—since a ring of the proper size could not be found in the local hardware stores. The assembly is shown prior to compressing the O-ring.

The driver is captive between the O-ring and the ¼-inch collar, and the latter is supported by the internal ridge of the coupling. Terminal posts, fitting phonocartridge-type clips, have been soldered to the electrical terminal board on the driver. The plastic-foam collar and the 1¼ inch cavity are part of the equalization system. The ¾-to-½ inch coupling is lined with plasticene (modeling clay) or epoxy putty to make a smooth transition to ½-inch tubing; this lining was formed with the help of a dowel rod that had been rubbed with wax.

A flare-fitting tubing cap of ¼-inch size has been cemented to the center of the perforated cover plate of the front side of the driver. Epoxy cement was used (room-temperature cure), sealing off the interior of the cap. The purpose is to eliminate a flexural resonance of the cover plate that causes a sharp response anomaly at about 9 kHz. Actually, the resonance is moved to about 1 kHz and its sharpness (Q value) increased. Its coupling to the acoustic field is reduced, and it becomes of inaudible consequence.

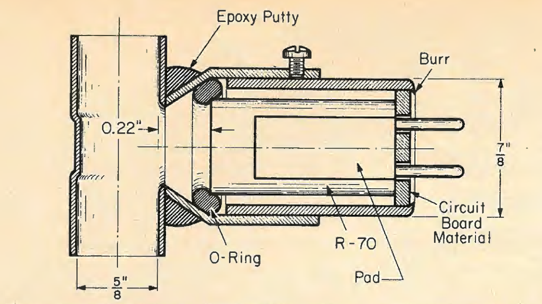


Fig. 4—Pickup mounting. The side branch of a half-inch tee has been removed (flush saw cut) and the tee cemented to a three-quarter-to-one-half coupling, from which the small end has been removed, and the joint reinforced with epoxy putty. With the O ring partially compressed, the cavity depth is 0.22 inch between the front of the R-70 microphone cartridge and the inside wall of the tee.

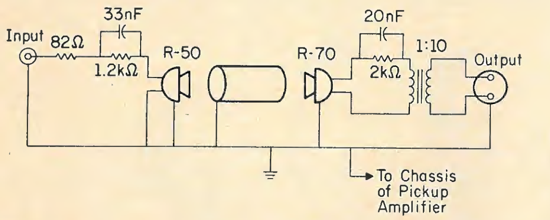


Fig. 5—Electrical schematic diagram of the wave propagation tube. The equalization networks are designed to give break points at 4 kHz (beginning of treble boost) and 16 kHz (end of boost) for an input source impedance of 120 ohms and an output load impedance of 47 kohms. The transducer impedances are 200 ohms. The grounding of the R-70 is optional, but the pickup-amplifier chassis grounding should not be made through a signal-bearing lead. Circuit-card mounting of the quarter-watt resistors and fifty-volt capacitors is used.

The arrangement for coupling the pickup microphone to the tube is shown in Fig. 4. A ½ × ½ × ½-inch tee fitting is used with the side branch removed by means of a flush saw cut. Also, a ¾ × ½-inch adapter fitting (C × C type) is used with the ½-inch end removed by means of a saw cut at the place where outside diameter starts to increase. These two parts are then joined by epoxy cement (the butt joint is cured at an elevated temperature) and subsequently packed on the outside with epoxy putty. The microphone seal is provided by an O-ring of ⅞-inch OD and 3/32-inch thickness. Using fittings of the NIBCO brand, the result is a cavity between the microphone and the long part of the tee that has a depth of 0.22 inch, and that can vary by ± 0.02 inch depending upon the deformation of the O ring. This cavity is also part of the equalization system.

The microphone holder consists of a 1¼ inch length of ¾-inch rigid copper tubing into which is fitted a ⅜-inch-wide strip of phenolic-fiber circuit-board material cut with rounded ends to make a snug fit. Ordinarily, it is good practice to remove from the tubing the internal burr left by the cutting tool, but the burr was allowed to remain at one end of this piece to hold this retaining strip. Two holes were drilled in the strip to accept the terminal posts of the R-70 cartridge. Phono-cartridge-type clips fit these posts well and are recommended for making the electrical connections. The holder also contains two pads made of ⅜ × ½-inch polyurethane foam gasketing material. These pads fit into cutouts in the R-70 frame, and their paper backing facilitates their sliding into the holder. These pads play no identifiable role in the equalization system but guard against the appearance of response anomalies.

The termination section of the wave-propagation tube is a 4.5-foot length of half-inch tubing into which has been pulled a tapered plug of polyurethane foam. This plug was cut, in sections, from half-inch-thick upholsterer's pads and bonded to string using ⅜-inch masking tape. Unless the tube is straight, this pulling operation is extremely difficult to perform properly. It was later found that a satisfactory termination can be made by cutting the termination tubing into 12-to-14-inch lengths and pulling a graded section of tapered foam into each. This is a much easier operation. Then, the graded sections of termination tubing may be joined in the proper sequence using ½ × ½-inch couplings bonded with epoxy cement. For such joints, the internally fitting member (tubing) has a small bead of epoxy applied near the end in a complete circumference. Then the insertion is made with a twisting motion so that this very sparing application of epoxy suffices to make a complete seal. Epoxy should always be very sparingly used, or else it will ooze into undesired places. The far end of the termination tubing is capped, and the cap is held in place with epoxy. A small hole is drilled in the cap to provide a slow air leak for pressure equalization.

Tapered polyurethane-foam pads are similarly fitted into the driver and microphone housings. These housings also are provided with slow leaks arising through the electrical connectors and unsealed joints. Nestled into this padding are equalization circuit boards and (for the microphone housing) a transformer.

The circuit diagram is shown in Fig. 5. The RC networks provide treble-boost equalization, with the low-frequency time constant at 40 μs (4 kHz), and the high-frequency time constant at 10 μs (16 kHz). The response is essentially level below 1 kHz and above 50 kHz, but rising in between with a total boost (both networks) of about 16 dB at 12 kHz. The rise beyond that point is largely inconsequential in overall effect, because of the steep cutoff of the transducers and the effect of the output-cable capacity. The ground return for the input circuit is connected to the tube, but the ground return for the output circuit is kept separate, except for one lead connected at the far end of the output cable, in order to avoid ground-loop hum. In spite of the epoxy joints, metal-to-metal contact is made throughout the tube, so that good electrostatic shielding is provided.

The input circuit is designed for connection to the headphone jack of a power amplifier. It is assumed that the maximum voltage output at the jack is about 20 V. rms behind 120 ohms, although it is inadvisable to permit this much voltage to appear in steady state at the higher frequencies, where the network provides rather little attenuation, since the long-term heating effects in the driver cartridge are not known. A few seconds of exposure to such levels appears harmless, however.

The output circuit is designed for connection to the microphone input jack on a preamplifier where the impedance would be 47 k ohms, shunted by 100 pF or less, including cable capacity. Reflected to the transformer primary, this would be 470 ohms, shunted by 10 nF or less. [The nanofarad unit (nF) is, of course, 1000 pF.]

Suitable microphone inputs are provided in the preamplifier section of many commercially-available amplifiers—offering a flat frequency response with sufficient gain to drive the power-amplifier section to full output from an input of a few dozen millivolts. Where such an input is not provided, a low-cost, low-noise preamplifier such as the Shure M-64 (with the 330 pF, input-shunt capacitors disconnected) may be used to drive a high-level input. In some cases, the M-64 provides an improved noise performance in comparison to an otherwise-available microphone input.

While the above output arrangements do provide satisfactory performance, improvements are possible through replacing the R-70 network by a treble boost obtained by modifying the feedback networks of the preamplifier to be used. When the R-70 drives the transformer directly, the bass response is improved, a better signal-to-noise ratio is obtained, and the top treble is less-easily affected by cable capacity. The modifications converting a PHONO input to a treble-boost TUBE input are set forth in Table 1 for the Shure M-64, the Dynaco PAT-4, and the Dynaco SCA-80. For the latter two, the SPECIAL input may be modified instead, upon suitable reinterpretation.

A low-cost arrangement could use two M-64's. The first would serve as the R-70 preamplifier, upon modification as in Table 1 and being fitted with output, level-set potentiometers (50 ohms), for which there appears to be sufficient panel space. The second would drive the power amplifier, and also embody the modification of Table 1, except that the capacitors would have the value 10 nF, and the bridging resistor would be 82 kohms, connected from S2 to a 100-kohms potentiometer with the wiper connected to the emitter end of R13 (R14 for the second channel). Thus, the second amplifier would provide a limited-range, tone-control action for adjusting the top end of the treble boost, while keeping the FLAT option available.

Design Considerations

Figure 6 shows a plot of the transmission (dashed curve) for such a tube (length of 16 feet and ID of 9/16 inch). The loss is calculated from the equation

$$A = 0.026\ n \sqrt{f}, \qquad (1)$$

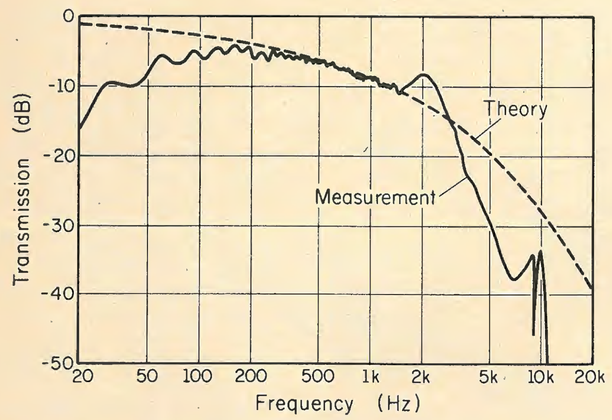


Fig. 6—Equalization requirements. The dashed curve shows the theoretical transmission response of the tube, and the solid curve shows the transmission with the transducers used, but without any of the equalization devices ultimately employed.

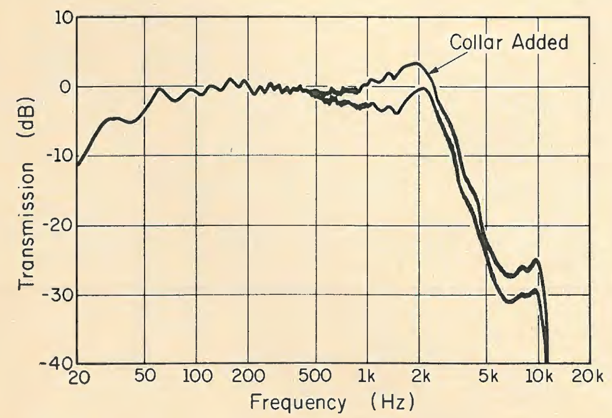


Fig. 7—Effect of the damping collar. The actual amount of high-frequency boost product by this means was adjusted to match the peak-supressing effect of the cavity to be placed in front of the R-50, for the peak of 2 kHz. More than 5 dB of boost was obtained. Here, and in Fig. 6 also, the response above 6 kHz was not very accurately measured. This test setup used straight tubing.

in which A is the decibel loss for air at 20° C, n is the length-to-diameter ratio of the tube, and f is the frequency in units of kHz [6]. For the present tube, the value of n is 341, for which the loss at 1 kHz is calculated to be 8.8 dB, while at 100 Hz and at 10 kHz, the values are 2.8 dB and 27.8 dB, respectively. Relative to the loss at 1 kHz, a gradual cut attaining 6 dB at 100 Hz and a gradual boost attaining 19 dB at 10 kHz would characterize the equalization demand.

This equalization demand seemed sufficiently severe to rule out the choice of a small-diameter tubing, e.g., one of 7/16-inch ID, for which the losses would be 30% greater. At the same time, a larger diameter seemed prohibited because of its bulk. A larger diameter would also begin to invoke the risk of multiple-mode excitation at wavelengths λ short enough to violate the condition

$$0.6\ \lambda > d, \qquad (2)$$

in which d is the diameter. For the present tubing, the risk is slight, since the tubing diameter is 14.3 mm, and the 10 kHz wavelength is 34.3 mm at 20° C, so that Eq. (2) is not violated until 14 kHz is exceeded, a frequency somewhat beyond the driver response capability. Thus, bends may be freely made in the tubing, and no anomaly is to be expected because of coiling the tube. In fact, it probably would be safe to insert 45° elbows in the tubing, and the possibility of an octagonal spiral layout, with such elbows coupling straight sections of rigid tubing, may be considered. Though such a layout could be extremely compact, it has not been tried.

Trials have been made of soft plastic tubing. Such tubing shows severe anomalous attenuation in the 1-kHz-to-5-kHz range. Although this attenuation may be partly lifted by packing the coil in sand, such tubing cannot be recommended, except possibly for the termination.

In lossless tubes, nonlinear propagation, in which the crests of the wave propagate faster than the nulls, can be obtained. The crests propagate with the velocity

$$c_+ = C_o + (1+\gamma)\ V/2, \qquad (3)$$

in which c_o is the sound speed, 343 m/s, the propagation speed of the nulls; γ is the ratio of specific heats, 1.4, and v is the particle velocity at the crests [7]. The percentage distortion in propagating the length of the tube is the percentage of a radian length by which the crests advance relative to the nulls. (Actually a distortion *index*, useful in estimation. For exact Fourier analyses, see Ref. 8.) This percentage is

$$D = \tfrac{1}{2}(1+\gamma)\ vT\ (2\pi/\lambda) \cdot 100, \qquad (4)$$

in which T is the propagation time of the tube, 14.1 ms in the present case. At 5 kHz, $\lambda/2\pi$ is 11 mm, so that ½ $(1+\gamma)$ vT should be 0.11 mm for 1% distortion. This corresponds to peak particle velocities of 6.5 mm/s. The intensity level into a specific acoustic resistance R is

$$I = \tfrac{1}{2} v^2 R. \qquad (5)$$

For the plane-wave acoustic radiation impedance (pure resistive) of 415 rayls, the level, in this instance, is 8.8 mW/m², or 99.4 dB referred to 1 pW/m². Since the area of the tube is 160 mm², the total acoustic power is 1.4 μW. If the transducer had an efficiency of 0.7%, the electrical power input would be 0.2 mW, represented by 0.2 V rms across 200 ohms.

The sensitivity of the R-70 microphone cartridge is known (within 1 dB depending on mounting) to correspond to an output of 1.8 mV (open circuit) for the above acoustic intensity. With tube loss and assumed driver efficiency, the overall insertion loss would be 50 dB. The measured loss of 49 dB confirms the figure used for the driver efficiency at 1 kHz.

For a lossy tube, the distortion is of reduced severity. Even without losses, the distortion increases only gradually with level,

and not abruptly as in a clipping-type overload. Thus, for peak velocities 10-fold greater (65 mm/s) the distortion would be only 10% at 5 kHz, and less than that with losses, whereas at 500 Hz, the nonlossy distortion would rise to 1%, etc. Generally, it seems safe to regard the levels calculated above to be suitable for ordinary soft-to-moderate passages, and to allow the 20-dB increase to correspond to the clipping point of the driving amplifier. At moderate frequencies, the network of Fig. 5 allows 2.5 V to excite the driver for an amplifier voltage of 20 V. At higher frequencies, more would be allowed, but most program material is rather weak at the higher frequencies.

The R-50 driver begins to produce noticeable buzzing sounds in the neighborhood of 10 V of excitation at about 200 Hz, so that it would appear that the driver power capabilities are reasonably well matched to those of the tube. At a peak particle velocity of 65 mm/s, the peak particle displacement is about 0.1 mm at 100 Hz. A diaphragm displacement of that magnitude would appear to be marginally safe, in terms of distortion, for a unit of the style of construction of the R-50 cartridge.

A dynamic-microphone cartridge was chosen as a driver, because it suits the power requirement, it is inexpensive, and its design already approximates that of a transducer suited for coupling to a frequency-independent radiation resistance. For this, it is desired that the response to an input force be an output velocity, and that this response be uniform in frequency over as wide a band as possible. For the tube driver, the output is particle velocity, and the input is the force developed by the current in the voice coil. For the microphone, the output is the diaphragm velocity (to develop the output voltage) and the input is the force developed by the acoustic pressure. It is seen that the design problem in the two cases is not quite the same, but it will be appreciated that the solution of the major elements of the problem in existing designs makes for a good starting point.

By contrast, an ordinary dynamic loudspeaker would make a poor starting point. Its decreasing velocity response above cone resonance does fit well with the increasing radiation resistance with increasing frequency, that is presented by free space to a baffled pist on, but not with a frequency-independent radiation resistance. Horn drivers would be better suited to the present purpose, but, apart from their cost and enormous power capabilities, they would also show a poor bass response in tube service. This would result from the stiffness needed to control the diaphragm motion below the frequencies (e.g. 300 Hz) at which the throat impedance of many horns becomes very small.

These basic considerations tend to show that a delay tube of moderate bulk can provide 14 ms of delay over a band extending up to 14 kHz with manageable equalization and a freedom from anomalous response. Moreover, high sound levels, making for a good signal-to-noise ratio, should be obtainable with acceptable distortion, using low-cost transducers whose design happens to be particularly apt.

(To be continued)

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Canby Gets Another Head

Edward Tatnall Canby

At last I've converted my weekly home-taped broadcasts to stereo. About time. But it wasn't easy to find the right tape recorder for the job. The one I have chosen to replace my professional grade mono equipment, which I assembled thriftily over the years, belongs in that interesting category that I like to call semi-pro. In terms of quality output, I now can do better in stereo than I ever did in mono. But the cost is far less, inflation or no. In my book, that is progress *deluxe.*

Taking over from my elderly but ageless Ampex 350 and its associated mixer control unit, both dating from the middle fifties and built like bulldozers—they're good for another fifty years at least—is a TEAC A-7030U stereo deck, top of TEAC's two-channel line. It is half the size of the Ampex and weighs perhaps a third, more graceful, less massive, more complex with its stereo facilities, and it probably won't last another fifty years. But will I need it for fifty years? I've decided to leave the Ampex to my grandchildren, if and when they are born, and it will surely be going strong when they grow up. Meanwhile, the TEAC is doing very well for me.

The background of this mono-to-stereo conversion is interesting. I went into full professional equipment, back in the fifties, because I had to. There wasn't anything else that would do the job. Moreover, I could swing the cost, with a bit of help from my friends. I grabbed the Ampex, slightly used, straight off a plane from a round-the-world tour, which had taken much less than 90 days. And when at the end of the tour the deck was declared surplus, it went right into my (puff, puff) arms.

I had a fancy control unit built to match, with pairs of phone inputs, microphone, master gain, and input level sets all over—so I could flip open the main pots all the way and come out exactly right. Vital if you have to talk intelligently while you do your own engineering. I also threw in a covey of equalizations, to cope with eccentric pre-RIAA LPs, and three separate kinds of input for each preamp, to cope with eccentric manufacturers of equipment. My microphone was a broadcast cardioid (Western Electric), for good speech sound and to help stifle studio background noise. I still use that one, despite having dropped it from a 10-foot boom onto a concrete floor a dozen years ago. The die cast frame broke but not the microphone elements. Rugged is the word for that kind of pro equipment!

And like all proper professional equipment, such as tanks, bulldozers, and subway cars, pro audio equipment is overbuilt to last and to take punishment. It is lovely stuff to have around the home—if you can afford it. I recently had the electronics in my Ampex replenished with all new tubes, resistors, and capacitors. They were the only really perishable items inside that sturdy frame.

Stereo, of course, greatly "enhanced" the cost of a good pro tape recorder, not to mention our subsequent inflation. Nowadays the dream of a nice home installation of that sort is impractical. We home folks are not corporations—where first cost is merely a minor part of the business operation. And as a matter of fact, we really do not need pro equipment. A good professional recorder runs 16-hour days for months and years on end, notably in the broadcast field. You may think you work your equipment pretty hard in your home, but I'll wager you don't keep *that* kind of a schedule. Nor do I. A five-hour session, capstan turning, is about what I can take, and that only once or twice a week, which is peanuts for pro equipment. In the home, or the part-time semi-professional operation, there is no need for such enormous ruggedizing nor for the massiveness and the bulk which it inevitably entails.

Imagine acquiring a new car that would run continuously with hardly a stop for more than 50 years, like a New York City subway car. That's your pro recorder.

And yet, there is a fascination to such equipment, a marvelous feel, a heft, a precision and purpose, which can spoil anybody who gets his hands on it. An enormous amount of pro recording goes on today and a vast number of the big machines are rolling. Inevitably, a lot of us get up close to them, especially the types who read this magazine, whether as engineers, producers, office boys, musicians, or just priviledged Kibitzers. Once hooked, you are ruined for *any* "consumer" tape equipment—those little boxes with the four-track stuff inside. It's almost embarrassing to use one after a heady day with a $25,000 16-tracker.

The intermediate machines, the semi-pro types, are the new way out, and a wonderful way it is. You *can* afford them, though it may hurt a bit. They aren't unduly ruggedized, and their size is moderate, even with big 10½ inch reels. But, definitely, they have a good measure of that silky professional feel and they will give you professional service for as many hours as *you* can take it. They are pro oriented but consumer built. You can feel the pro aspect at the first touch of the controls, and, let me tell you, that is a good feeling.

The little boxes, of course, can turn out tapes that meet any professional specs you want to name. In important ways they are just as versatile as the big ones, in their consumerish fashion. But there is a difference. Oh, yes! If you don't think so, then an intermediate is waiting for you. You'll go for those big reels, that silky, satinized metal, the flashy big buttons and the all-solenoid controls. You'll flip at the really *fast* forward and rewind, the effortless instant stops and starts, the controlled jockeying of tape between reels, the quiet alacrity of all the motions, the total absence of tape snafu (well, almost total), the truly automatic reel-end arrest and the neatly managed quick shut-offs when a splice comes apart or a tape breaks of its own accord. You'll wonder at the swing arms that cushion tape shocks, the big, generous heads (not two but three, or four or more), the huge pro VU meters (especially these!) which read all functions, not just some, and do it precisely. Mouthwatering.

Yet beyond these fairly typical semi-pro features there are differences. Extraordinary. And here is my caveat, my warning. Look before you leap! Good quality you will find in a wide range of such machines. You can take it for granted. But specific operations, features, approach, can vary wildly. You must match your machine to your particular needs or you may be in trouble. I spent huge quantities of time snooping around the industry, trying tape decks at hi fi shows,

To see Mr. Canby with his additional head, readers are referred to the Stanton Magnetics advertisement, which appears on page 17 of this issue.

quizzing the promoters and designers, before I found what would work for me. A machine that might be ideal for you could be useless in my studio.

Take editing, for instance. My radio program depends on it. Most people don't bother, except for a few beginnings and endings. To edit, you must pull your tape out of the recording slot, leaving the reels in place, then put it back. You must *see* the playback head and be able to mark the tape at the right spot. That means getting a marking pencil in there. Some otherwise excellent machines wholly ignore this aspect of taping. Tape threads *into* the slot very nicely but you can't get it *out* again, except by running all the way through. Or it catches infuriatingly on all sorts of projecting bumps and knobs and pressure pads, until you're ready to scream with frustration.

I'm not joking. One otherwise superb machine, I found, simply will not allow you to remove the tape from the slot—*unless you take off the capstan.* So I remember from a hi fi show demonstration, though I now find it hard to believe. And in many machines, it is utterly impossible to mark the tape; no room. Yet some of these machines are excellent sellers and top values in their field—for those who do not edit or who do their splicing on a different machine.

Design bias leads the semi-pro recorders in many interesting directions. The well known Crown line, with its various models, are supposedly (rumor says) designed for the use of missionaries in the field. Some missionaries! The Crown philosophy is to provide for every conceivable need, to meet every known situation and all with top quality. They do cost a bit more, but not much, all things considered. Astonishingly dense ranks of switches and knobs and volume controls. Meters that read everything, and switches to control them. A "computer" that controls the tape motions with batteries of clicking relays. Wow! For the right user, such a machine is a dream to play with or work with. For me, it is too complicated. I tried to do a broadcast or two and gave up. I couldn't think fast enough. For instance, to change from playback to record, I had to move *six* switches, channel by channel separately, and all different, and I had to look to be sure not to land on a wrong switch position. I can do that operation on my new TEAC with one hand, instantly, and blind. The two channels are hooked together for all normal actions.

The TEAC A-7030U, as a matter of fact, is clearly modelled on the classic Ampex configuration, updated: the same swing arms, with the right-hand one controlling tape motion or stop; the same inertia idler pulley on the left, to stabilize the tape movement, and the same open-faced heads, minus pressure pads. The TEAC's head cover comes right off and there the heads are, nicely exposed with nothing to get in the way, though a overhanging frame makes me squint a bit. Easy to mark the tape, and it lifts free with no obstructions at all to be laid on the nearby editing block. Is *that* important to me! But it may not be for you.

I particularly like the solenoid motion controls on the A-7030U. They are grouped together compactly under the fingers of one hand and you play them lightly, like keys on a piano (which is nice for a musician such as me). The single RECORD safety button (not two of them) is right within reach of a finger on the same hand, and it works only from a dead stop. Good idea; no mistakes. (On other machines you must work two RECORD switches, and you can often do it while the tape moves). TEAC starts and stops instantly and effortlessly, with a lot more precision than my old Ampex, and thus saves me hours of time. It stops so quickly that I can often do my cueing without bothering to go through the "rocking" procedure. Just stop, right on the exact spot. It is a pleasure to "drive" this baby, I assure you. Its motions are smooth, efficient, and time-saving.

Yes, you can break a tape. (I always try this, just to see.) Push the REWIND or FAST-FORWARD, then STOP and before the motion ceases, push PLAY. *Snap!* But the pro machines will mostly do the same, including Ampex. No machine is totally foolproof, what ever the claims. (Look out for Mylar. It doesn't snap; it stretches into a tight tube and there goes your recording.) Occasionally, my TEAC control buttons go dead for a moment, then come to life. Same thing for my Ampex 350. Relays not relaying. One learns to live with them.

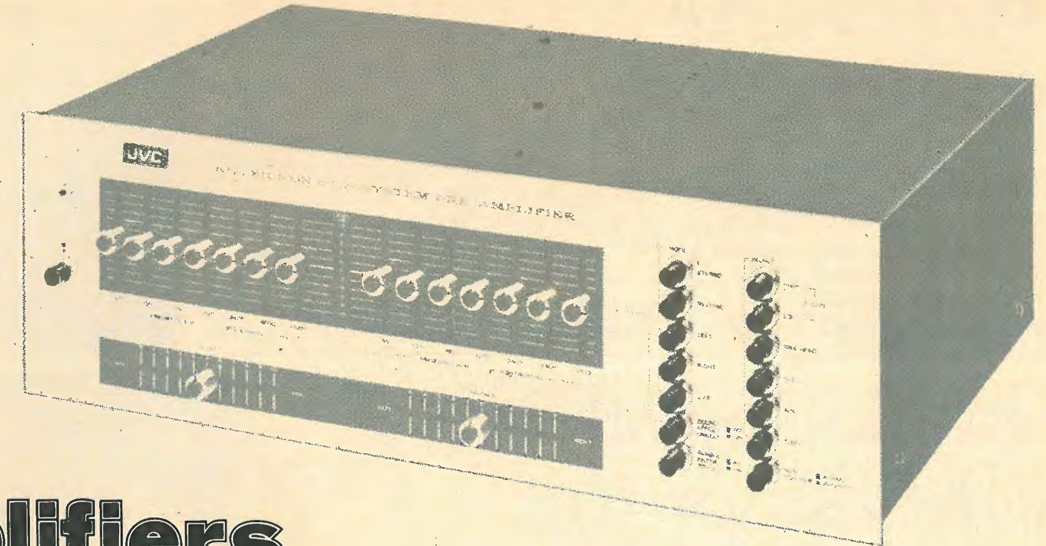
The controls on this TEAC are marvelously simplified, allowing full independence of the two channels when needed, but hooking them together for all normal stereo functioning. A single button shifts from PLAYBACK MONITOR to INPUT MONITOR, like the single RECORD button; another throws in the alternate four-track playing head, using the same controls. No worries about proper recording curves; they change with the speed change. The motor, as in Ampex, goes off when tape is slack, allowing for much valuable rest time during long editing or recording sessions. (In many machines, the capstan turns all the time.)

Except for microphones, all connections go to the rear and are out of the way. The TEAC capstan is quite large, excellent for good tape contact, a vital feature if speed is to remain exact under varying reel loads. Reel holders, useful for vertical operation, are quick and easy to remove. (Some machines require much screwing of fasteners, an annoyance.)

One or two mild complaints. Aesthetics today require that all good looking machines have sharp edges and corners on the metalwork and on buttons and arms. The TEAC is good to look at but the edges do snag tape more easily than the old-fashioned rounded shapes on the Ampex 350. You can't argue with fashion, I suppose. Tape tangles under the TEAC's swing arms more easily than on the Ampex, perhaps a matter of clearance underneath. But the Ampex has a circular slot around its spindles through which tape sometimes gets into the motor well with disastrous results. Not so for the TEAC. The microphone inputs are badly placed, right below the volume controls where your hand bumps into the connectors. (This is one thing TEAC could easily fix.)

The feature that most agreeably caught my eye in the TEAC literature seemed to have been designed just for me by Japanese telepathy—a built-in mixer with inputs for microphones *and* high level signals and a pair of two-channel split friction volume controls. You work the channels as one, but you can adjust each to any possible combination. In one instant, this eliminated the need for that extra special mixer unit I used with the old Ampex. No level set nor phono preamp, but a handy old Dyna control unit gave me exactly what I needed there, its own level control a perfect level set. Also extra inputs for other sources, including a second tape recorder, plus stereo balance setting, stereo-to-mono mix and so on. Mild problem with my old microphone, not enough gain. I feed it to both channels for AM-FM listening. The TEAC provides a modern 10,000 ohm input impedance and there's a mismatch of sorts I suppose. More modern microphones match up better, but I merely talk a little louder, that's all.

An interesting follow-up to the above discussion occurred as I was writing this piece. My New York City stereo broadcasts were temporarily discontinued because my tapes, done on the TEAC, slow down on the air, my voice sounding older and more tired until I become at least 90 and the music sagged dismally flat. My fault? Not on your life! The large capstan TEAC checks right on speed, as I know from a careful strobe test and recordings made at the start and end of the reel. The trouble is with the expensive professional transports used at the station. I hear tell that they have a very small, high speed capstan, and they won't play certain tapes, including mine. Probably too slippery. (I use 3M tape.)



JVC Model 5011

Amplifiers

A look at requirements and specifications

George W. Tillett

MANY YEARS ago, in 1880 or thereabouts, Lord Kelvin said, "I often say that when you can measure what you are speaking about and express it in numbers, you know something about it. When you cannot express it in numbers, your knowledge is of a meager and unsatisfactory kind." Very true—but if you cannot understand the numbers or their significance, then your knowledge is *still* unsatisfactory. Technical specifications of *any* kind can be very confusing to the beginner, and those relating to amplifiers are certainly no exception. Indeed, the multiplicity of standards and the trend of some manufacturers to use artificial power output figures, for instance, make translation quite difficult for engineers—let alone beginners.

Before we take a look at the figures and see what they mean and what really matters, it will be as well to list the basic requirements of a stereo amplifier—and here we will include separate preamplifiers and power amplifiers, combined or integrated units as well as the amplifier sections of receivers. Here they are, not necessarily in order of importance: Low distortion, wide frequency range, high damping factor, adequate power output, unconditional stability, low hum and noise levels, low crosstalk, good channel matching, and adequate tone controls, filters, and general facilities. Add to these, reliability and ease of handling plus good styling.

Distortion

The perfect amplifier does not yet exist, in spite of the claims to the contrary, and even the best, the most expensive, do not reproduce an exact replica of the input signal—99.99% perfect, —well perhaps! Both tubes and transistors are inherently non-linear devices but special circuits and the use of negative feedback can reduce the distortion to insignificant amounts. Distortion can be divided into two kinds—harmonic and intermodulation, although as far as amplifiers are concerned, both are somewhat interrelated. Harmonic distortion means the generation of spurious harmonics of the input signal. Thus if a pure 50 Hz note is applied to a poor amplifier, it would come out as an amplified 50 Hz fundamental plus a certain amount of 100 Hz second harmonic, 150 Hz third, 200 Hz fourth, and so on. Figure 1(a) shows a distorted waveform with a high second harmonic content. As a matter of interest, an amplifier that had *only* second harmonic distortion might not sound too bad. As is well known, all musical instruments depend on the production of harmonics (or partials) for their characteristic tone color or timbre. This is why a note played on a flute, for instance, sounds different from the same note played on a clarinet or trumpet. Moreover, the relationships of harmonics and fundamentals determine the tone quality of such instruments as the Violin or piano, so if our amplifier only changed those relationships, the results would not necessarily be *that* unpleasant. Indeed, it might merely convert a grand piano into an upright or vice versa (see Fig. 2). One thing is certain however, the high-order harmonics—the 5th, 7th, 9th, etc. which are not harmoniously related to the fundamental in the musical sense—are objectionable even if present in small proportions. Figure 3 shows how harmonic distortion is measured with a wave analyser. This instrument is tuned successively to the harmonics of the test frequency—usually 400 or 1000 Hz and the readings are expressed as follows:

$$D = \frac{\sqrt{E_2^2 + E_3^2 + E_4^2 + \cdots}}{\sqrt{E_1^2 + E_2^2 + E_3^2 + E_4^2 + \cdots}} \times 100$$

where D = percentage of total harmonic distortion,
E_1 = amplitude of fundamental voltage, and
E_2 = amplitude of second harmonic voltage, etc.

Because high order harmonics are progressively more unpleasant, many authorities advocate a "weighted" distortion factor in which the harmonics are weighted in proportion to their relative unpleasantness [1]. Obviously it is very difficult to assess relative unpleasantness and so the weighted system is not often used. A simpler method of measuring harmonic distortion is by using a bridge type of instrument that balances out the fundamental, leaving all the spurious harmonics to be measured as a total percentage. Figures obtained by this method are listed in specifications as *Total Harmonic Distortion (THD)* at a specific input frequency, usually 1000 Hz.

Some of this material was previously published in the British Audio Annual (Hi-Fi News) 1965, to which due acknowledgements are made.

Intermodulation Distortion

This is the production of spurious sum-and-difference frequencies when two or more frequencies are passed through a non-linear amplifier. Figure 1(b) shows a 50 Hz and a 1000 Hz note applied to an amplifier which we will label B. It is a reasonably good amplifier and so the resultant waveform is fairly pure. However, if we apply the same frequencies to amplifier A, it is quite a different story. The distorted 50 Hz note deforms the 1000 Hz waveform by intermodulating it as shown in Fig. 1(c). In this process it creates a whole series of sum-and-difference frequencies—1050, 950, 1,100, and so on. These spurious frequencies play havoc with the reproduction of music as they are not harmonicly related to the original tones. Most of the harsh distortion associated with overloading and Class B crossover distortion is due to intermodulation or IM. There is no simple relationship between IM and HD but with well designed tube amplifiers it is normally 4 : 1. In other words, 0.25% HD roughly corresponds to 1% IM—up to overload point. There is no correlation for transistor amplifiers where the IM is largely dependent on crossover distortion. Many of the early transistor amplifiers had a significant amount of IM distortion that actually increased at low output levels (see Fig. 4). Crossover distortion tends to produce more IM than harmonic distortion—moreover the higher order harmonics predominate. This brings us to the big questions: *How much distortion can we actually hear,* and how much is acceptable? Various tests have been made in the past and the general conclusion is that a THD of 0.5% at 1000 Hz can just be detected. Under domestic conditions it is possible that most people could tolerate a rather higher amount—especially if one considers the limitations of most program sources. But, bearing in mind the subjective unpleasantness of the high-order harmonics and the fact that we do not usually know their percentages in the quoted figures, it is best to play safe. A good amplifier, then, should not have a greater distortion than the following: 0.3% at 1000 kHz, 0.6% at 40 Hz and 1% at 10,000 Hz. Most specifications only give the 1000 Hz figures but if the distortion rises appreciably at the ends of the spectrum, it will normally show up in the IM figures. Standard IM frequencies are 40 and 7000 or 70 and 7000 Hz in a ratio of 4:1 and IM distortion should not exceed 1.5% at any level up to rated power output.

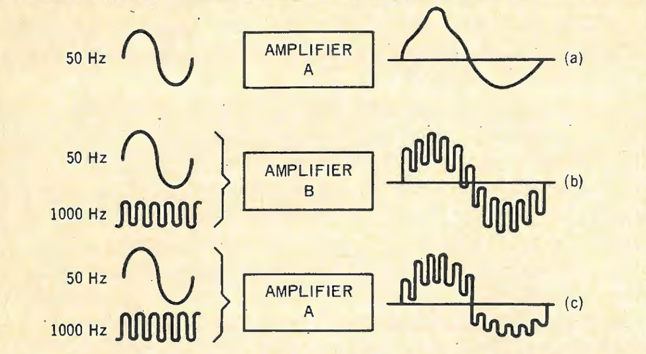


Fig. 1—Harmonic and intermodulation distortion.

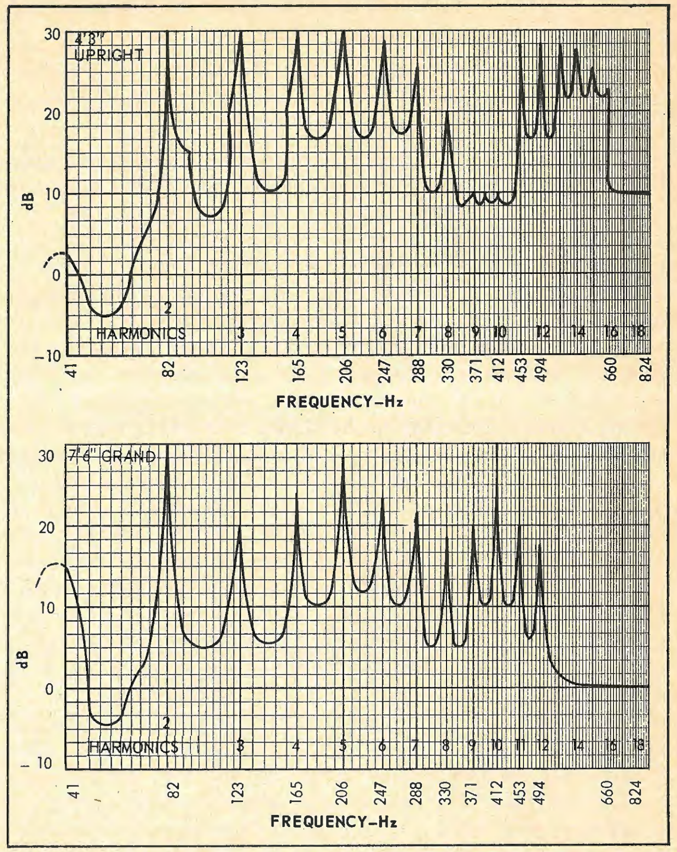


Fig. 2.—Harmonics of E3 (41.2 Hz) produced by two pianos using equal hammer velocity. (From "Pianos, Pianists, and Sonics" by G. A. Briggs, Cahners Publishing, 221 Columbus Ave., Boston, Mass. 02116.)

Fig. 3—Wave-analyzer method of testing amplifier distortion.

Frequency Response

What should the frequency response be? I would say 5 Hz to 70 kHz within 1 dB for a really good amplifier, but the bandwidth could be reduced to 20 Hz - 30 kHz within 3 dB without losing too much "fi." Seventy kHz might seem a little high for the roll-off point but many authorities believe that the overall bandwidth ought to extend up to 200 kHz or higher. Others say, "Wideband amplifiers are strictly for the bats." What is the truth about the matter? Well, there are several reasons for the high upper limit. Here are some of them:

1. The internal bandwidth of an amplifier using a large

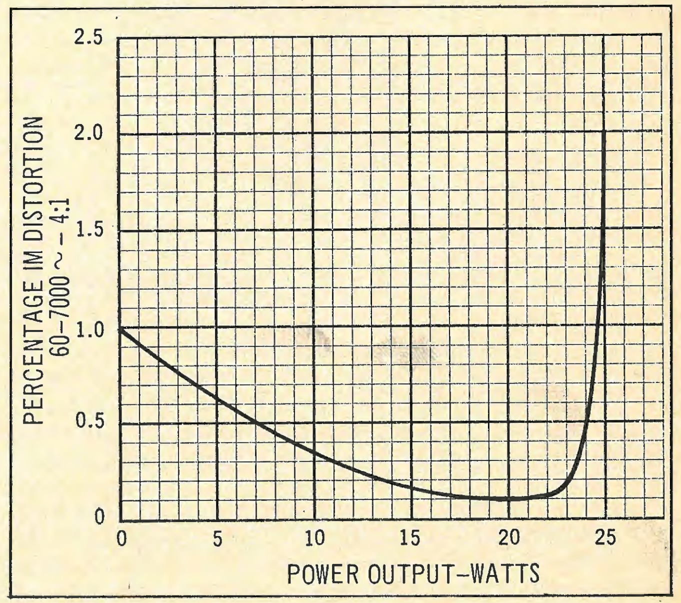


Fig. 4—Showing intermodulation distortion characteristic of a typical early transistor amplifier.

amount of negative feedback must extend to at least one octave above the audio range to maintain stability,

2. Assuming a flat response up to 20 kHz is required, then the rate of attenuation above this frequency must not exceed 6 dB per octave to avoid "ringing," and

3. To reproduce square waves properly the pass-band must be ten times the input frequency. The difficulty here is to equate square waves with music and some experts maintain that it is only necessary for an amplifier to pass square waves up to 4 kHz, thus indicating a response up to 40 kHz. There is a school of thought which goes much further than this [2]. Their argument is based on the Helmholtz theory that most music really consists of a series of tiny transients that blend together in our ears, and they claim that an amplifier must therefore be able to reproduce a square wave of 20 kHz without distortion. This means a bandwidth of ten times 20 kHz or 200 kHz. One reason why this point of view is not widely accepted is the limitations in program sources but one thing is certain—the higher frequencies play a more important part in fidelity of reproduction than was thought possible some years ago. Tests have been made indicating that the removal of frequencies above 20 kHz or so can be detected by listeners who are deaf to anything above 15 kHz. One such experiment described by Slot of Philips involved two scientists, one of whom was absolutely deaf to frequencies above 10 kHz and the other above 11 kHz [3]. Both could unfailingly tell when a filter was switched into circuit that brought the highest limit down from 18 kHz to 12 kHz. Slot goes on to say, "It may be that the highest frequencies, even though they are not actually observed, still give rise to intermodulation within the hearing system and that the absence of these subjective intermodulation tones is considered unnatural."

Experiments carried out some years ago at Wharfedale with very wideband amplifiers were not conclusive—sometimes listeners preferred a response up to 200 kHz, sometimes a more restricted range was considered more natural—depending on the program material. It would seem logical to assume that if the upper frequencies *do* contain some kind of information, then they could also add to the sum total of distortion. In other words the wider you open the window, the more dirt blows in. So it would seem essential that a very wide band amplifier should have a switched filter to give best results. Summing up, an overall frequency range (measured at 1 watt) of 5 Hz to 70 kHz within 1 or 2 dB really represents a middle-of-the-road approach, with 20 Hz to 30 kHz within 2 dB being the lower limits. Incidently, frequency range should always be quoted with a dB reference, the often used formula "frequency response 2 to 50 kHz" or whatever being virtually meaningless.

Power Output

How much power is needed? What with experts recommending anything from 5 to 500 watts and speaking about sine-wave power, Music Power, peak power, and so on, it is no wonder many people are a little confused. Ample power should be available to handle peak transient signals without distortion and how much is enough depends on the size of the room, speaker efficiency, and personal taste (and possibly that of the neighbors). Speaker efficiencies range from the 25% of the large horn systems right down to the meager 0.5% of the small bookshelf models. So 1 watt fed to a large horn system would make as much noise as 50 watts into a bookshelf unit. A good average figure is 3% increasing to 5% for larger floor models. A total power of 30 to 70 watts would be needed for a small to average size room, increasing to 100 watts total for a large room. These figures refer to old fashioned rms watts, not Music Power. What exactly *is* Music Power? According to the IHF it is "the greatest single frequency power that can be obtained without exceeding the rated total harmonic distortion when the amplifier is operated under standard test conditions, except that the measurement shall be taken immediately after the sudden application of a signal and during a time interval so short that supply voltages within the amplifier have not changed from their no-signal values." In practice, this means using an external stablized power supply.

Obviously, if the d.c. voltage remains constant, then the Music Power will be the same as the rms, sine wave, or continuous power. On the other hand, if the power supply is badly regulated, then the Music Power rating will be a lot higher than the rms figure. Put another way—the power supply is so designed that it will only deliver full voltage for a very short duration, so if the signal is not a sine wave but merely consisted of transient peaks lasting a few milliseconds, the supply voltage would remain constant.

This is a very comfortable theory because it enables considerable economies to be made—not only with the power supply transformer, rectifiers, and capacitors, but with the output transistors and their large expensive heat-sinks. More than this, it enables an unscrupulous manufacturer to double or even triple the apparent output figures. But of course, there are snags—for one thing there is no agreement as to how *long* the music transient peaks shall be. Peak powers of some organ works like Bach's *Toccata and Fugue in D* are quite long in duration and the demands of some kinds of electronic music are even more stringent. The only safe guide is to judge an amplifier's power performance by the rms figures and to bear in mind that the Music Power rating should not exceed the rms figure by more than 30%. The specified power should apply when both channels are driven. Note also that power output is usually greater for 4 ohm than for 8 ohm loads.

That ± 1 dB Rating

Sometimes used in specifications as "Power output, 100 watts ± 1 dB," "± 1 dB" is a dishonest method of rating, as we have often pointed out. It is roughly equivalent to 21%, so that the 100 watt amplifier only need put out 79 watts!

Overload

An amplifier should have a smooth overload characteristic with almost instantaneous recovery. A poorly designed power supply not only fluctuates in d.c. voltage according to the signal, but produces severe distortion near overload point due to a superimposed sawtooth "hum" waveform caused by insufficient smoothing at high currents.

Power Bandwidth

This can be defined as the frequency range that lies between the extremes where the available power falls by half or 3 dB. Figure 5 shows the power response of two amplifiers, one giving half power at 15 Hz and 45 kHz and the other (Amplifier A again) at 55 Hz and 8.5 kHz. Note that both give their rated power at midband and so both could be sold as 50 watt amplifiers. These days when output transformers are as rare as acoustic phonographs, loss of power at low frequencies is not too common but reduced output at the very high frequencies caused by transistor limitations is occasionally found. How important is power output at high frequencies? It used to be thought that full power was not necessary above 8 kHz or thereabouts. But with improvements in program sources and sound equipment as a whole, it has been found that the upper limits ought to be a good deal higher. For instance, the upper partial or harmonic of the cymbals at 18 kHz has an amplitude equal to the 680 Hz fundamental and this instrument can still put out considerable power at 25 kHz. [4]. To be on the safe side, the half-power point should not be much lower than 30 kHz. (Some of the best transistor amplifiers can deliver half power to well over 50 kHz and full power up to 20 Khz.)

At the lower end, full power should be maintained down to at least 30 Hz although some authorities say that 40 Hz is a more reasonable figure. It is true that the lowest fundamentals of some musical instruments such as the piano, contrabassoon, and harp (believe-it-or not) are below 40 Hz but as a general rule the second and third harmonics are greater than the fundamental. However, there are no great problems in designing solid-state amplifiers to give full power at the low end, and so in practice a good amplifier will have a power bandwidth of 15 to 30,000 Hz at 3 dB.

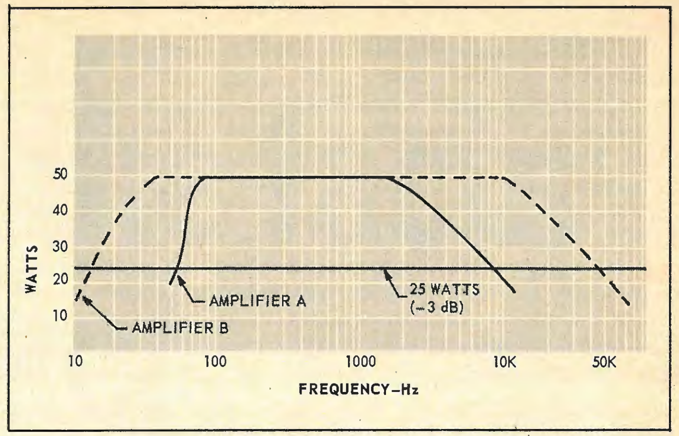


Fig. 5—Power bandwidth of two amplifiers.

Damping Factor

This is the ratio of load impedance to the amplifier's internal impedance. A damping factor of 50 which is a typical figure for modern amplifiers, means that at the nominal output impedance of 8 ohms, the output resistance would be 8 divided by 50 or 0.16 ohms. A high damping factor means that the loudspeaker "sees" a low resistance which tends to damp any tendency for the diaphragm to vibrate at its natural frequency or resonance. It might be thought that the higher the damping factor, the more efficient this acoustic brake becomes, but in practice any increase above 20 results in little further improvement. This is because the resistance of the speaker voice coil is effectively in series with the output. Long speaker leads will also have an effect, as will crossover coils.

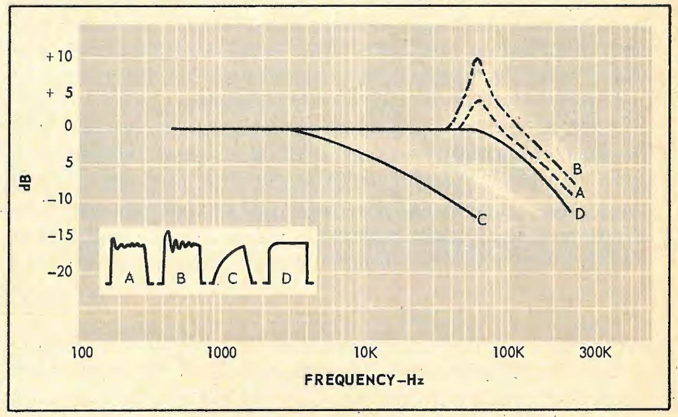


Fig. 6—Responses with 10 kHz square wave resolutions.

Transient Response

Transients are usually defined as sounds of short duration such as made by cymbals, piano, and other percussion instruments. However, they are also produced by many other instruments at the start of notes. Indeed these sounds play a significant part in stereo location [5]. It is fairly well known that amplifiers must have a wide frequency response to reproduce transients properly, but there are other factors involved. The response must be free from peaks and supersonic oscillations which can cause severe transient mutilation or "ringing." Most amplifiers these days use a high amount of negative feedback, and it is essential that the high frequency response is carefully controlled with the necessary phase correction.

Square-wave signals are used for testing and Fig. 6 shows four response characteristics: A shows a slight overshoot or ringing due to a small peak at about 60 kHz and B shows the effect of a larger peak; this type of spurious oscillation may be continuous or it might be triggered off by an input signal or "switching transient." Curve C has a restricted high frequency response which shows up as a rounding of the square wave, and D has a wide frequency response giving excellent square wave resolution with very slight rounding. Amplifier B would certainly sound "edgy" and harsh, while C would sound dull and lacking in "attack." Amplifier A—the one with the small peak—would be more unpredictable in how the ringing might be modified by the speaker load; it may improve slightly or it might even be triggered into an incipient oscillation like amplifier B.

Transient response can also be adversely affected by some types of tone control, steep-cut filters. As far as specifications are concerned, few makers specify square wave rise time which indicates both transient response and bandwidth to some extent, but 3 microseconds at 10,000 Hz would be considered excellent (See Fig. 7). In any case, a 1000 Hz square wave should be reproduced without rounding or overshoot (with tone controls in the "flat" or cancel position). Above 7,000 Hz the waveform will show appreciable rounding—depending on the overall frequency response. Sometimes inductive-capacitive loads are used for testing (no loudspeaker behaves as a pure resistance), and slight overshoot at higher frequencies is permissible under these conditions.

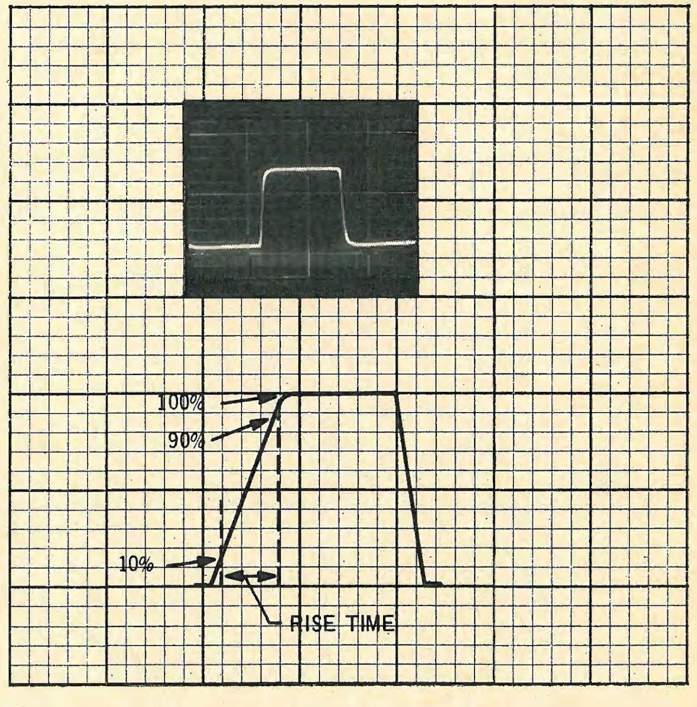


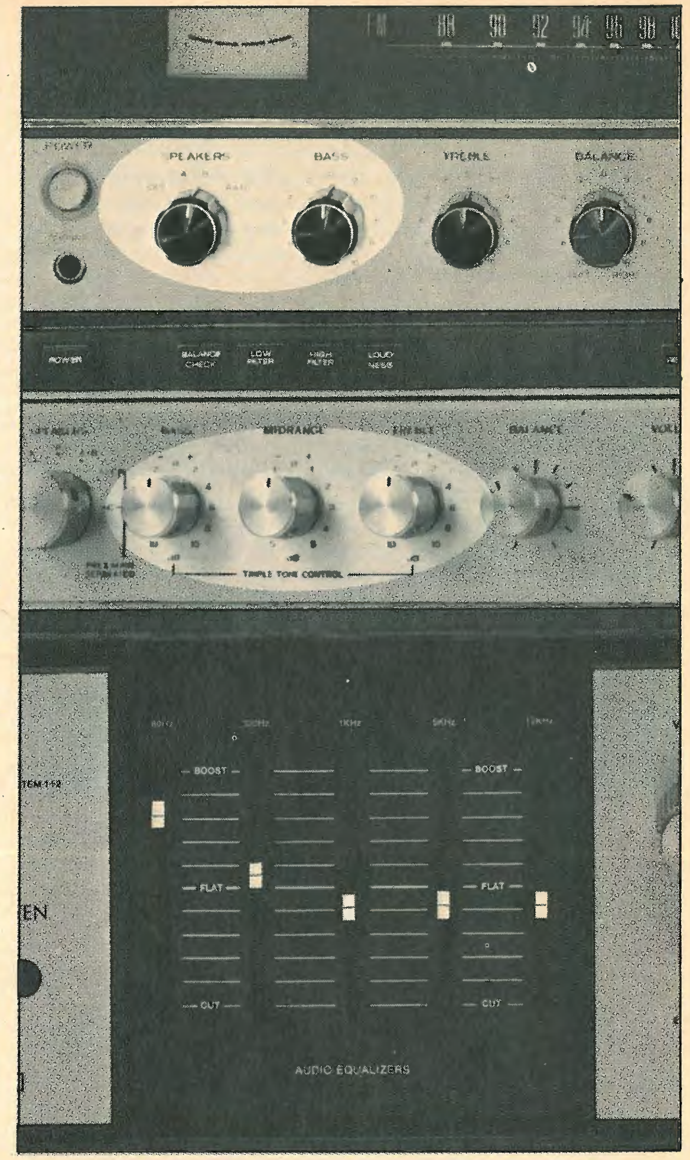
Fig. 7—The square wave rise time is the time taken to achieve 90 percent resolution from 10 percent.

Stability

Not often mentioned in the specifications but important for all of that is stability. A really good amplifier should be unconditionally stable, which means that there should be no signs of instability with any combination of inductive and capacitive loads. Because of the high amounts of negative feedback used to reduce distortion, the stability margin on some amplifiers might not be very great. Consequently, severe load conditions such as those caused by complex crossovers might result in sufficient phase shift to cause positive feedback or instability. In practice, the only complex load likely to be encountered would be electrostatic speakers, and readers who are contemplating the purchase of such speakers should check that their amplifier is suitable.

Tone Controls

Dealing now with the preamplifier section we come to the tone controls. All control units have bass and treble controls—some have bass, middle, and treble and some divide the spectrum into even more ranges. The intention is to provide means of compensating for studio acoustics, recording deficiencies, room acoustics, and so on. For normal use a lift and cut of 10—12 dB at 40 Hz and 10,000 Hz is more than adequate. Two types of control are in common use, the passive type and a feedback type originated by P. J. Baxandall. The passive type has the effect (see Fig. 8) of rotating the response about a central hinge, usually 1000 Hz but the Baxandall type works in a different manner as the boost and cut are initially confined to the ends of the scale (see Fig. 9). This has the advantage that the lower bass frequencies can be boosted without appreciably affecting the 300 to 500 Hz region. At the treble end, the Baxandall would function like a filter in the cut position, but on the other hand you could not get a lift from the mid-range if so desired unless you turned the control almost fully up. So, both methods have their advantages and disadvantages. The addition of a mid-range control makes for greater flexibility—so do the more complex 4 and 5 unit controls. Whether such refinements are worthwhile is a matter of economics and personal choice. In most amplifiers the tone controls are ganged so both channels are controlled together but independent controls are provided on a few more elaborate amplifiers.



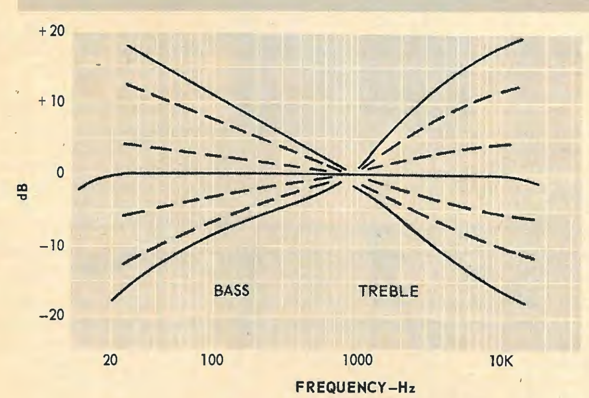


Fig. 8—Typical passive tone control characteristics.

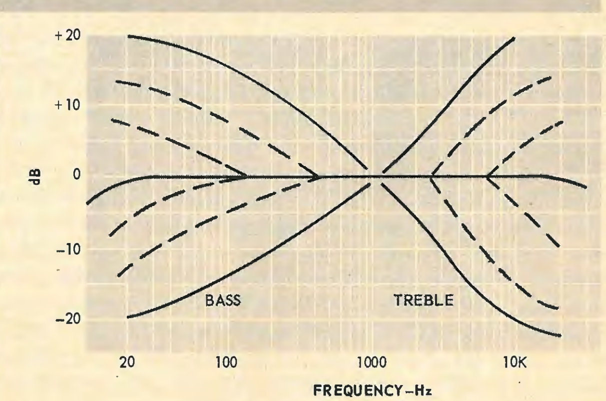


Fig. 9—Baxandall tone control characteristics.

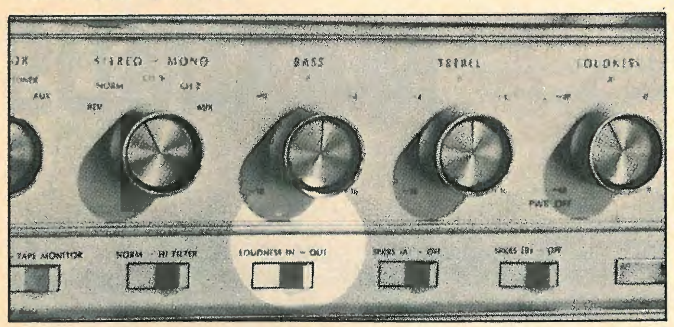
High Frequency Filter

In order to reduce distortion produced by imperfect broadcast transmissions, tape hiss and those "barbed wire strings" favored by some recording companies, it is sometimes necessary to restrict the high-frequency response by means of a filter. Most of this distortion, especially pickup tracing distortion, affects the higher frequencies or upper harmonics, and the usual tone controls cannot always deal with it without destroying too much of musical value (throwing out the baby with the bath-water). In its simplest form then, a filter might consist of a single switched unit operating from about 7 kHz although sometimes a choice of two frequencies such as 4 kHz and 8 kHz might be given. Many European amplifiers are fitted with more elaborate filters giving a continuous attenuation from 4 to 20 kHz—plus another control which can vary the actual *rate* of slope.

Low Frequency Filter

Sometimes termed a RUMBLE FILTER as this is its main function. It has been said that the rumble filter is a sign of defeat. Perfectly true, but it may well be the best practical way of dealing with the problem. Even quite expensive transcription motors produce *some* rumble, which is accentuated by stereo pickups that are, of course, sensitive in the vertical direction. A speaker system that is very efficient at the low end could show up rumble but it must be emphasized that rumble at subsonic frequencies can produce considerable IM distortion by overloading the output stage or speaker system although it cannot be heard as a specific sound. Some pickup arms have what is in effect a built-in rumble filter. (Some have resonances that could make matters worse!) But in any case, a low frequency filter is desirable. If it is switched, a frequency of 40 Hz would be a good choice but if the filter is built in, then 20Hz would be a better compromise. Most filters merely consist of a switched capacitor so in order to arrive at a reasonable attenuation, the roll-off point has to be about 200 Hz—so we have another case of throwing that baby out with the bathwater!

Loudness Control



This originally was a separate level control that automatically lifted bass (and to a smaller extent treble) at lower volume settings. These days it refers to a switch that changes the circuitry to permit the normal volume control to perform those functions if so desired. At one time, much controversy centered about the need for such controls. Advocates refer to the Fletcher-Munson intensity curves which prove that the human ear is non-linear and that more power is needed at the low frequencies to produce the same apparent loudness when listening at low volume levels. Most readers are probably familiar with the Fletcher-Munson curves, which are shown in Fig. 10. Note that the curves are relative and that they are almost parallel above 2000 Hz. Now the theory is this: Suppose a symphony orchestra produces an intensity of 90 phons, and (out of consideration for the neighbors) we wish to reproduce this at a level of 70 phons, we apply compensation based on the difference between the appropriate level curves as shown in Fig. 10 and everything is fine. But is it?

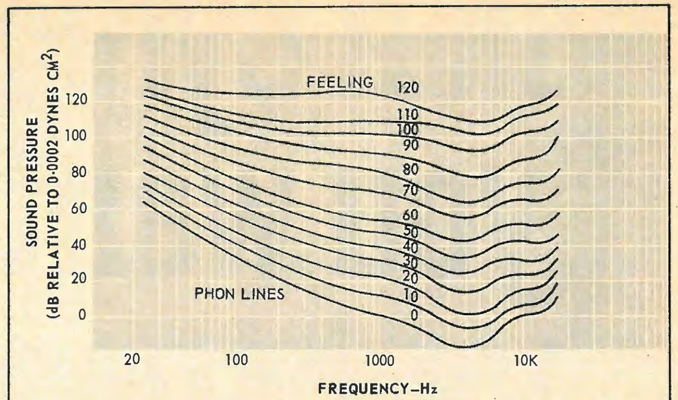


Fig. 10—Fletcher-Munson equal loudness curves.

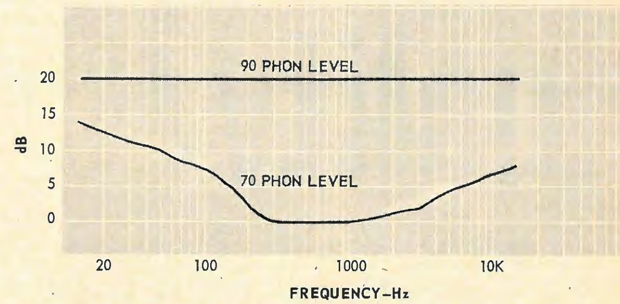


Fig. 11—Relative level curves.

Critics say that the Fletcher-Munson curves were taken with sine waves and because of the effects of intensity on pitch and harmonic structures, they cannot apply to music. Moreover, we listen to live music at various intensities without being worried by bass deficiencies. They go on to ask, "Does an orchestra *really* sound that unnatural when we listen some distance away?" It is possible that the desire for bass lift when listening at low levels is a psychological one but the situation is complicated by the fact that some speaker systems, particularly reflex types and very small bookshelf models, are non-linear and sound a little thin at low volume levels.

So is a loudness control necessary or not? My own opinion—and I stress that is a personal opinion—is this: If you have a really good speaker system it is doubtful if you will feel the need for such a facility. On the other hand, it does not cost much to provide and you need not use it if you do not want to! Another personal opinion: It really ought to be called a "softness" control!

Hum and Noise

A combined figure is usually quoted for hum and noise. The hum component consists of a mixture of power line frequency (60 Hz here) plus some second and other harmonics due to smoothing deficiencies, power transformer radiation, etc. The noise or hiss is due to random impulses generated by tubes, transistors, resistors—in fact any component passing current. It is almost aperiodic in form, that is, it is not confined to any particular frequency or bands of frequencies. In practice, there is an identifiable low-frequency component caused by transistor current variations.

Specifications usually quote the figures for hum and noise in decibels related to the full rated amplifier power output. The disadvantage of this method is that a large amplifier can have identical noise figures to a smaller one but the background would be higher. Thus a 10 watt amplifier might have a measured noise of 10 millivolts at the output. This is one thousandth of 10 watts or 60 dB. But a 100 watt amplifier with the same 10 millivolts of noise would have a noise level of -80 dB referred to full output! Some makers use a

1 watt reference figure, but if the full power figure is used then this must be borne in mind when making comparisons. The important question is: *How much hum and noise can we accept and how much is inaudible?* Here it is difficult to give a precise answer, as a lot depends on the proportions of hum and noise, the frequency response of the amplifier, response and sensitivity of the loudspeaker system, room acoustics, ambient noise, and other factors. Noise can be accentuated by peaks anywhere in the system, particularly in the speakers, and a spectacular improvement can often be obtained by replacing the treble unit with a better one having a smoother response. This will, of course, not only reduce background noise but it will have an effect on tape hiss and record noise too. As a general guide, hum and noise should not exceed the following:

Input	Sensitivity	Hum and noise unweighted (rel. to 1 watt)
Tape Head	2 mV	50 dB
Phono	3 mV	50 dB
Radio	100 mV	70 dB
Tape	200 mV	76 dB

Preamplifiers are usually rated with respect to the output voltage which would give figures some 20 dB higher than those above.

Sensitivity

There is not much to be said about sensitivity–the figures in the table above are fairly typical. Due attention must be paid to the overload factor as the amplifier input stages come before the volume control. Regarding the question of signal-handling capacity, we find a very interesting situation. Phono pickups are usually rated at so many millivolts per cm/sec, thus the Empire 999 is rated at 2 millivolts cm/sec. Sometimes an average figure is given, e.g. the output at 7 cm/sec but obviously the output at the *maximum* recorded velocity will also be important. When it is realized that these peak velocities may reach 30 cm/sec it will be seen that the amplifier will have to cope with 60 millivolts from the 999, and there are some cartridges that give an even higher output. To be on the safe side, then, the amplifier input stages must handle a signal of at least 80 mV. This was not too difficult with tube amplifiers but certainly posed problems for early transistor amplifiers using low voltage "front-ends." However, the advent of high voltage, low noise silicon transistors eased the situation considerably, and some of the best amplifiers can handle signals up to 100 millivolts.

A few words on impedance: A value of 47 K will suit the majority of pickups and this has become the accepted standard. Few amplifiers have tape head input sockets these days and one of the reasons is the difficulty of correct matching. Tape heads have widely differing characteristics and there is also the question of tape speed. As a rule then, tape equalization provided by a conventional amplifier is very much a compromise and it is much better to use a matched, equalized preamplifier inside the recorder and feeding the output into the amplifier at high level.

Channel Matching

Most likely cause of deviations from accurate channel matching are the ganged controls, particularly the volume control. Figures are rarely given but a good amplifier should maintain channel matching within 2 dB at all control settings. The usual range provided by the balance control itself is plus and minus 6 dB, more than enough to take care of divergencies from program sources and ancillary equipment.

Crosstalk

Crosstalk between channels posed some problems with tube amplifiers due to the high impedances involved but most transistor amplifiers have at least 30 dB separation up to 10,000 Hz or higher.

Facilities

Some amplifiers boast a great variety of facilities such as headphone sockets, phasing switch, provision for center channel, oscilloscope display, extension speaker switch, speaker mute, arrays of indicator lamps, and so on. Apart from the headphone socket which is really a "must" the rest are refinements which of course cost money and you have to decide whether they are worth it or not. Switched inputs for tuner, phono, and tape are more or less standard but it is definitely useful to have an extra auxiliary position for a cassette recorder or an AM radio. Two switched phono inputs are provided on some amplifiers–useful for comparison purposes but probably not *that* important for most people.

Reliability

It is no use having an amplifier with all the desirable features if it continually gives trouble. So put *reliability* at the top of the list. . . . Here is where transistors score heavily over tubes which, in theory, begin to deteriorate from the moment the unit is switched on. The most likely cause of trouble is the accidental shorting of the speaker terminals or abnormally high input transients which could damage the output and possibly the driver transistors. Fuses are only a partial protection (they do not blow fast enough) but the majority of amplifiers use protection circuits that limit current or switch off the d.c. power supply under overload conditions.

Well, we seem to have covered everything–except for the question of styling. This is very much a matter of personal taste, but whether you prefer a simple, uncluttered control panel or one more like something from an Apollo space rocket it should be functional. Knobs should not be too small to handle properly, switches should be positive and if knobs are inscribed with some indication of position, these indents must be near the panel marks to avoid parallax errors. Inscriptions should not be too small or use unduly fancy typeface so a magnifying glass is necessary to read it and knobs ought not to be too close together.

A recent trend is the use of slider units for some of the controls–usually for tone and balance. One advantage is that the actual positions can be seen from some distance away but many people find the old fashioned rotaries a little easier to use. It's a matter of opinion, because slide controls are used almost exclusively in recording studios–although they are invariably mounted in a horizontal plane. A final point concerns on/off switches: At one time it was always considered good design practice to mount these separately to avoid undue wear on the volume control. This is not so necessary these days and a combined control certainly helps to avoid a disconcerting blast of sound as soon as you switch on. . . .Good Listening! Æ

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Doppler Anyone?

Henri A. van Hessen*

I DON'T believe in Doppler distortion, chiefly because I'm not sure I ever heard it. I think it's a bogey, brought to life whenever developments grind to a halt.

But what should Doppler *distortion* sound like? All we hear from the experts is that its effects are more pronounced with smaller speakers. Well, we all know that bigger speakers sound better than little ones, but also that good little ones sound better than nasty big 'uns. Is this Doppler? I doubt it.

Then, when you get so foolish as to actually try to follow the experts in their reasoning, you find that they all seem to live in a different world from the one we find ourselves in. I have drawn a set of diagrams to illustrate what I mean.

Fig. 1 shows the basic Doppler generator on which they all seem to base their calculations. It is a circuit constellation forcing the loudspeaker to produce intermodulation. To achieve this, it does not depend on any inherent fault of the speaker. On the contrary, the better the speaker, the more pronounced will be the Doppler effect. All you have to do in order to get your sums right is to take care that the two outputs do not get the chance to work back into the other generator.

We can only regret that this measuring set-up has nothing, but nothing, to do with listening to music at home. That is a very different set-up, as outlined in Fig. 2.

The orchestra has been simplified to two loudspeakers, each of which is now fed from its own generator. The result now is an acoustical mixture of two pure tones. No Doppler in Room A. (If there were, there would be Doppler in the concert hall and we would not have to worry about our speakers, would we?) The microphone will pick up the resultant sound mixture and it should be evident to anyone who does not call himself an expert, that the electrical signal given off by the mike is not identical to the mixture as generated in Fig. 1. This is brought about by the fact that while picking up the higher frequency, the microphone diaphragm is also moving at a lower frequency. The micriphone will therefore produce a signal in which the two frequencies have modulated each other. This, however, is no reason to worry, as we shall see in a minute.

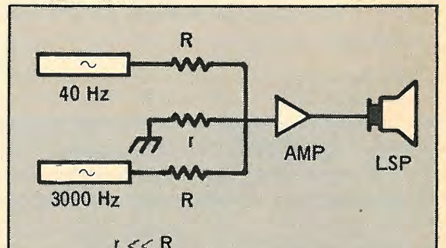


Fig. 1—The laboratory set-up, a basic Doppler generator.

FIRST of all, let's get rid of the black box, X. X represents the convolutions the signal is subjected to before it reaches the listener at home. It has no bearing on our reasoning whether the signal is put on tape, transmitted on FM, or processed into a record, as long as it is not distorted too much. We can disregard the whole X procedure for the purposes of discussing Doppler.

Now let's go and listen in Room B, which could be your own living room. Any Doppler, dear? Well honey, frankly, I'm not going to have Doppler today, thank you.

The reason you are not bothered by Doppler is that you do not get the electrical signal to which the experts have applied their immense powers of reasoning. You do receive a pre-Dopplerized signal, and if your loudspeaker is any good, it will just de-Dopplerize it. Why it has to do this is illustrated in Figs. 3 and 4.

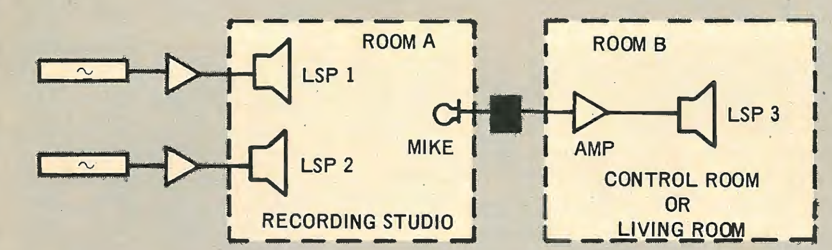


Fig. 2—The listener's situation. Lsp 3 is your own loudspeaker.

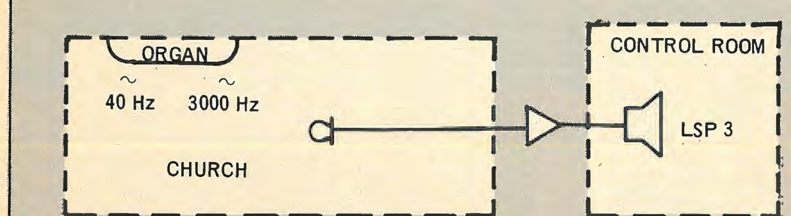


Fig. 3—The recording process simplified—step one.

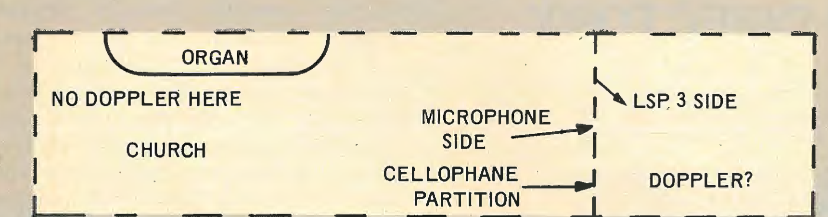


Fig. 4—The ultimate simplification.
(Exactly where did we lose Doppler distortion?)

In Fig. 3 we have done away with the two speakers, 1 & 2, shown in Fig. 2, which were immaterial anyway. They are replaced by organ pipes giving out the same two frequencies. If we agree that we have no Doppler distortion in the church, there will be no Doppler distortion in the control room—which still could be our own living room. To clarify this statement a bit more, we shall simplify our diagram once again, coming to Fig. 4.

I think we are all agreed that the amplfier plays no part in the Doppler proceedings, all the experts putting all the blame on the loudspeaker. Indeed, all the amp can do if it is a good one, is to see to it that the loudspeaker diaphragm is moving exactly in step with the microphone diaphragm. This it does, since it is paid to do this, with an accuracy of over 99.99%. (If it's a good amplifier, of course.) We can therefore leave the amplfier out of the picture alltogether, so that we can paste the two diaphragms together. What remains, is a church partitioned by a very thin sheet of cellophane. I am waiting for someone to step forward and explain how there is no Doppler distortion on the left, and Doppler on the right side.

Frankly, the whole Doppler discussion reminds me of a man who finds himself able to calculate to several decimals how great a centrifugal force is exerted on the stylus by the spinning record, driving the arm to the outside. Silly, you say? There isn't any such force at work? Then, what do you call the Doppler advocates? Æ

*Henri A. van Hessen is a Dutch engineer and writer, well known in Europe, who contributes regularly to European audio journals.

Behind The Scenes

(Continued from page 14)

seconds providing that high frequencies are suitably attenuated (since high frequencies tend to give directional information).

Now the engineer is ready to adjust the variables involved, and he is faced with an intimidating array of as much as nine faders, eight equalizers, and must deal with several values of reverberation time and time-delay. Mr. Eargle states, "The difficult adjustments are the ratio of *delayed* and *reverberant* components in the rear channels as well as the equalization of those signals. The ratio of delayed and reveberant sound determines the effective 'distance' of the rear channels from the front pair, while the equalization would be adjusted to simulate the color of reverberation most compatible with that already present in the original recording. The notion of scale factor is important. Settings of the variables which may be appropriate for processing a large scale symphonic recording will almost invariably be inappropriate for a string quartet or a chamber orchestra." Mr. Eargle also wisely points out that the entire "direct-reverberant" concept for re-processing more intimately recorded program material is questionable, and another tack is indicated. I have mentioned many times that since a string quartet can be physically accomodated in the average living rooms of today's homes, recording such a group with reverberation must be very carefully done if intimate musical values are to be preserved.

To anticipate an obvious question. . . can consumer-type spring-reverberators be used to simulate four-channel sound? The answer has to be a somewhat qualified yes. They afford only one color of reverberation with a fixed decay rate and this is a decidedly limiting factor. Most objectionable is their characteristic metallic "Boinnggg" sound, which immediately spotlights their artificiality. The Eargle process four-channel stereo tapes are the result of controlling the variables of time-delay, reverberation color, and reverberation decay rate.

Well, there you have it. . .a method of signal processing that can make available a vast library of utterly convincing, ultra-realistic four-channel stereo tapes, which can then be used "as is," or be used as a master for encoding discs. Mr. Eargle is to be congratulated for a technical *tour de force* and some very original thinking. I'll end with the words of that familiar ballad, "This Could Be The Start Of Something Big!" Æ

Equipment Profiles

KLH Model Forty-One Stereo Tape Deck

MANUFACTURER'S SPECIFICATIONS:
Speeds: Three—7½, 3¾, and 1⅞ ips. **Heads:** Three—erase, record, play. **Reel Size:** 7". **Motor:** Synchronous. **Wow and Flutter:** 0.15% at 7½ ips. **Signal-to-Noise Ratio:** 60 dB without Dolby, 68 dB with Dolby circuit in. **Inputs:** Microphone, high-level line, low-level line. **Output:** Line, 1000-ohm impedance. **Dimensions:** 14¼" wide, 11½" deep, 7½" high overall **Weight:** 23 lbs. **Price:** $199.95.

It was bound to come sooner or later—a quarter-track stereo recorder with built-in Dolby circuitry and well within the means of any potential tape recordist. And if you do a lot of tape recording, the money you can save by recording on this machine at 3¾ ips instead of the usual 7½—money saved in the cost of the tape used—might possibly pay for the recorder itself.

The Model Forty-One is a compact unit which incorporates a single Dolby processor which is switched in or out at the user's discretion, and which can be used for both recording and playback, depending on the mode of use. Thus while the deck has three heads, it is not possible to monitor the signal on the tape during the recording operation. However, once the deck is connected to a typical receiver, monitoring is continuous whether recording or playing back. During recording, monitoring is off the source and while playing back it is, of course, off the tape. The advantage of this arrangement is that it is not necessary to do any switching on the receiver or preamp as you change from recording to playing back. Furthermore, the use of separate heads for the two operations makes it possible to use heads most suitable for each. It is well known that the playback head should have a shorter gap than is common on the record head.

The Forty-One has a single bar lever which controls all tape motion. And it has one feature which is both unusual and desirable—there is a mechanical delay when switching from fast forward to play so it is not possible to break the tape. Similarly, when going from the rewind position to play, one must first pass the stop position. The lever has five positions—rewind, stop, play, pause, and fast forward, so it is possible to depress the record buttons, operate the lever to play, thus locking in the recording action, then to pause, which stops tape motion but leaves the unit in the record mode for adjusting levels. This is not an unusual feature, but it is effective and convenient to operate.

There are separate input-level controls for each channel in the recording mode, permitting balancing of the channels, and in addition there is a master record-level control which affects both channels at once, thus retaining the balance originally set with the individual controls. The playback level control is located on the underside, and operates on both channels at once. The three operating speeds are selected by a lever between the two reels, and equalization and speeds are switched simultaneously.

Line output phono jacks are located on a small panel on the bottom of the unit (when used in the horizontal position), as are two pairs of line input jacks—high and low levels. As mentioned previously, the output level control is on this same panel. There are no protective fuses in the unit—either in the line or d.c. circuits.

For those who would use the deck in the vertical position, there are two plastic "feet" provided which have adhesive strips that are applied to the wood cabinet at specified places, and two reel clamps are provided for use when the deck is in the vertical position. Microphone input jacks are accessible through an opening in the right side of the cabinet and are the usual phone jacks.

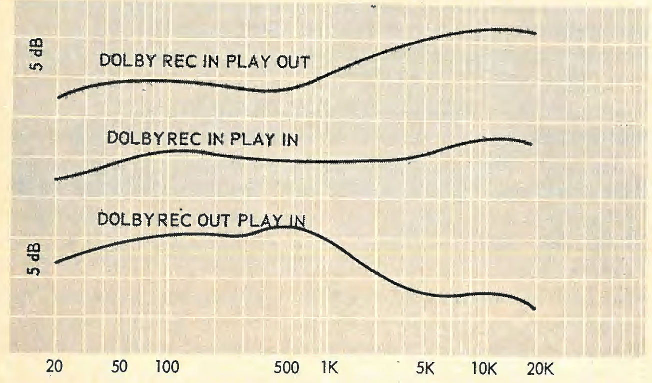
Circuitry

The KLH Model Forty-One is neatly built with one master printed-circuit board fitted with six sockets into which are plugged the individual circuit modules—one preamp section for each channel, one Dolby processor for each channel, a record amplifier section for both channels, and the erase/bias oscillator. Adjustments are provided for record level, playback level, and meter level to conform to the requirements of the Dolby system. These settings are factory-adjusted and are not intended as operating controls for the recordist.

The preamp module consists of two transistors with feedback to adjust for playback at the three speeds, along with a booster stage, with the master level control followed by the individual-channel level controls between them. The booster stage is followed by the chanel-selector switches and feeds the source signal to the line-output jacks. The record signal is fed to an emitter follower with a bias trap in the emitter circuit, and its output feeds the Dolby processor which employs five bipolar transistors and one FET, along with five diodes, and the control voltages are switched from play to record to effect the proper action in the unit. In the record mode, the high frequencies below a certain level are boosted a specific amount, as shown in Fig. 1, which shows the frequency response of the record circuit at 20 dB below "0" level as indicated by the VU meters, along with the frequency response of the processor in the play mode, also at 20 dB below "0" level. Since these two curves are complementary, the resulting playback is essentially flat over the entire frequency range and over a wide range of levels. Thus the unit adds pre-equalization to the recording and adds complementary post-equalization in the playback, but only to those signals below a given level in the source. A similar effect could be obtained by boosting the high frequencies in recording and reducing them in playback, but one would run the risk of overloading. The net result of the entire Dolby process is that the noise or hiss level is reduced by about 8 dB in low-level passages so the hiss is practically inaudible during the quite portions of the music—which is the only time it bothers the listener, since it is masked by the program material itself at high levels.

The record signal leaves the Dolby processor and is fed to the record output stage which is a single-ended push-pull unit. From there it is fed to the record head through a bias trap, after which the bias signal is added before feeding the record heads. The erase/bias oscillator module employs two transistors operating in a push-pull circuit. The erase signal is fed to the erase heads from a tap on the oscillator transformer, while the bias signal feeds from the secondary of the oscillator transformer through potentiometers for adjusting bias, and thence through small capacitors to the record heads.

Fig. 1—Response curves run at different Dolby conditions, as indicated. These are included to show the effect of the Dolby processor.



Performance

We made the usual playback-from-standard-tape measurements, with the results shown in Fig. 2, then followed by measurements of the record/play responses. In order to evaluate the effect of the Dolby system, we made recordings at 20,15, and 10 dB below the indicated 0 levels at all three speeds in all possible combinations—record Dolby in, play Dolby in; record in, play out; record out, play in; and record out, play out. These responses are shown in part in Fig. 1 (space does not permit showing all 24 curves, so only the 7½-ips curves are shown and at only the -20-dB levels).

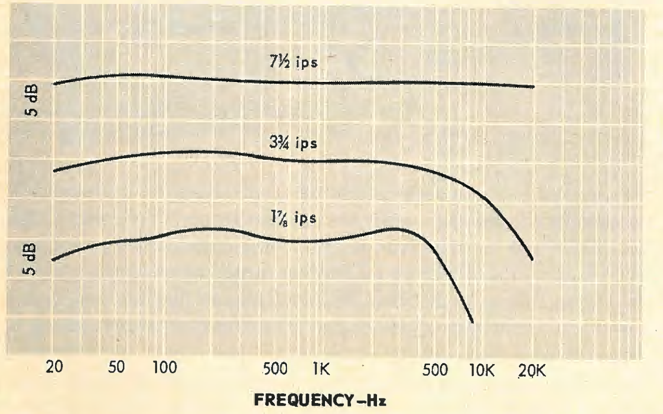


Fig. 2—Frequency responses at the three speeds from standard tapes. Dolby circuitry was switched out for these curves.

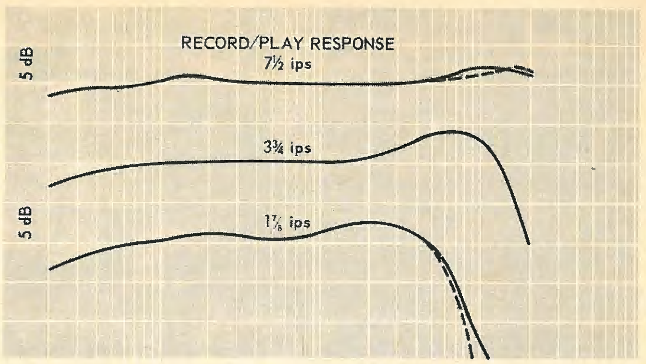


Fig. 3—Record/play responses for the three speeds. Dashed lines show deviations between Dolby-in and Dolby-out conditions.

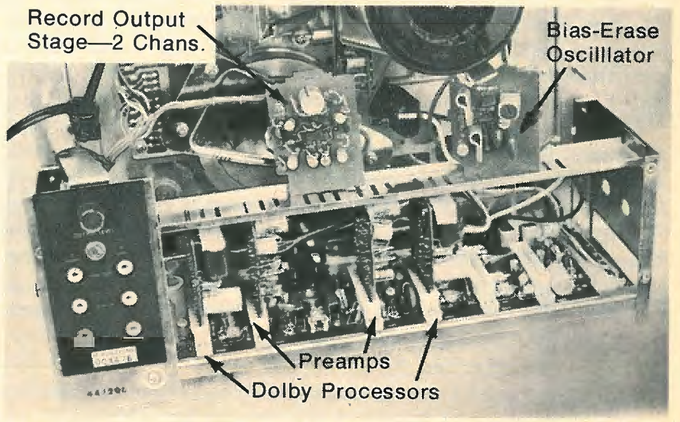


Fig. 4—Showing two of the printed circuit assemblies removed and placed on locking strip. Board at right is the 105-kHz bias-erase oscillator panel; the other is the record output state for both channels.

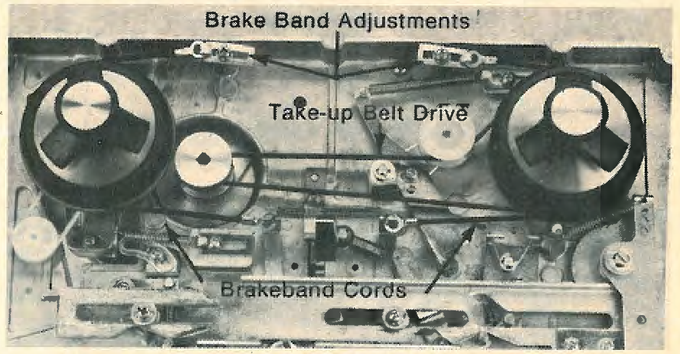


Fig. 5—The two reel turntables of the Model 41, showing the take-up belt drive and the adjustable cord brake bands.

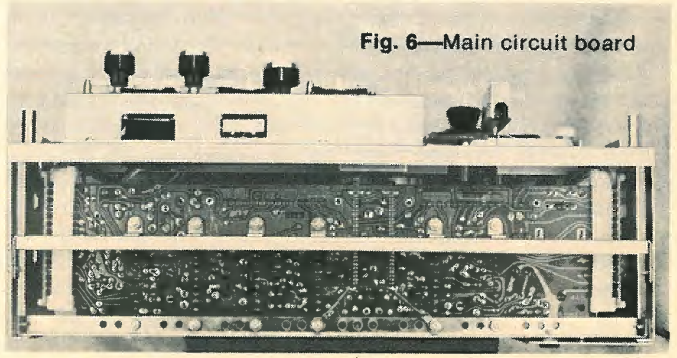


Fig. 6—Main circuit board

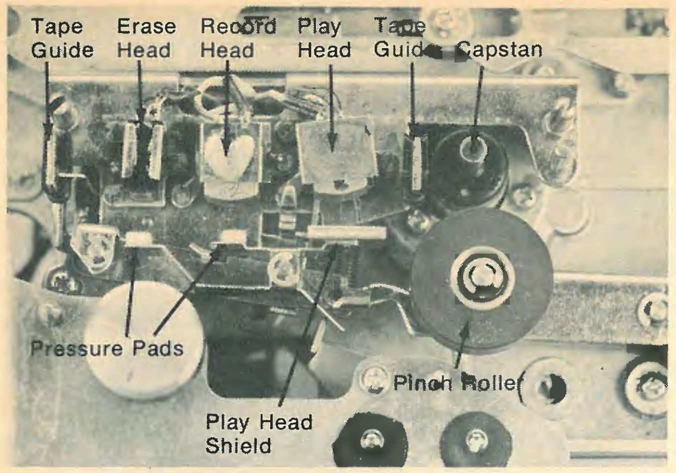


Fig. 7—Close-up of the head assembly showing the pressure pads and the shield for the playback head. Tape is well guided past the heads.

These curves show how the Dolby circuit functions to produce essentially the same frequency response after processing as it does with no processing whatever, either in recording or in playback. Note that the frequency response is flat within ± 1 dB from 40 to 20,000 Hz at 7½ ips, and flat within ±4 dB from 40 to 13,000 Hz at 3¾ ips. At 1⅞ ips—which is not recommended for music recording in the instruction booklet—response is within ±4 dB from 40 to 7000 Hz—still a creditable response.

We measured distortion at the indicated "0" levels at all three speeds at 1000 Hz, and found 1.2 per cent at 7½ ips with the Dolby circuit in, 1.3 per cent with the Dolby out; at 3¾ ips, distortion measured 1.3 and 1.5 per cent, and at 1⅞ ips figures of 1.3 and 2.0 were obtained.

Next we measured distortion at 10,000 Hz and obtained figures of 1.1 and 1.8 per cent at 7½, and 1.6 and 5.0 at 3¾. We did not measure the distortion at 10,000 for 1⅞ ips, but topped out at 5000 for this speed, where we found 1.0 and 3.2 per cent. At the two higher speeds, we found an increase in distortion at 5000 Hz, measuring 2.2 and 2.5 per cent at 7½ and 2.8 and 3.2 per cent at 3¾. Distortion at 100 Hz was below 2 per cent for all three speeds.

Wow and flutter measurements at 7½ ips gave figures of .05 per cent in the range from 0.5 to 6 Hz, 0.11 in the range from 6 to 250 Hz, and 0.12 in the overall—0.5 to 250 Hz. Similar measurements at 3¾ ips showed figures of .06, 0.11, and 0.14 for the same ranges, and at 1⅞ ips the corresponding figures were 0.11, 0.13, and 0.17 per cent—all being in the excellent category. The bias oscillator frequency is nominally 105 kHz, and measured on this particular machine at 101 kHz. Waveform was satisfactorily pure—always an aid in keeping hiss to a minimum.

The input signal required for "0" level indications on the VU meters with the record level controls at maximum was 350 mV at an input impedance of 330 k ohms for the high-level line inputs, 104 mV at an impedance of 100 k ohms for the low-level line inputs, and 0.5 mV at an impedance of 1000 ohms for the microphone inputs. Line output level for a signal recorded at "0" level was 2.2 V with the output-level control at maximum.

All measurements were made with Scotch 111 tape, which is the type recommended for use with this machine. Low-noise tapes would require a change in the bias setting for satisfactory operation. The 3-per cent distortion point was found to be at a level 5 dB above the indicated "0" level, and the signal-to-noise ratio measured 63 dB with the Dolby in and 56 with the Dolby out—an indicated improvement of 7 dB. Fast forward and rewind times measured 160 seconds for 1200 feet of tape.

The unit employs 27 transistors, 2 FET's, 11 diodes, and one Zener which serves as a reference for the regulated supply.

We made a lot of recording and listening tests with the Model Forty-One. With the Dolby circuitry switched out, the unit performed much like any other tape recorder. With the Dolby switched in, we noticed a definite improvement in the background hiss, particularly at 3¾ ips. Thus the KLH Model Forty-One permits the recording of much material at 3¾ ips using the Dolby system and does so with considerably reduced tape hiss. And that is the reason why so many recordists are interested in the Dolby process. *C.G. McP.*

Pioneer Model TX-900 AM/FM Stereo Tuner

MANUFACTURER'S SPECIFICATIONS:
FM SECTION: IHF Sensitivity: 1.7 μV. S/N: 70 dB. **THD (Mono):** 0.3%; **Stereo:** 0.5% **Alt. Channel Selectivity:** 65 dB. **Image Rejection:** 95 dB. **Capture Ratio:** 1.5 dB. **Frequency Response:** 50-15 kHz + 2 dB. **Antenna Inputs:** 300 ohm balanced and 75 ohm unbalanced. **Stereo Separation (FM):** 38 dB @ 1 kHz; at least 30 dB from 40 Hz to 8 kHz. **Output Level:** 1 volt. **AM SECTION: IHF Sensitivity:** 10 μV. **Image rejection:** 70 dB at 1 MHz. **Antenna:** Built-in Ferrite Loopstick. **Output Level:** 1 Volt @ 30% Modulation. **GENERAL: Dimensions:** 15 15/16 in. w. x 5 9/16 in. h. x 14 in. d. **Power Requirements:** 12 watts maximum @ 117 V., 60 Hz. Weight: 17 3/16 lb. Suggested Retail Price: $259.95 (includes metal cabinet).

The first impression one gets in examining this "top-of-the-line" tuner from Pioneer is that it is large—as large, in fact, as some of the more recently released complete receivers of other manufacturers. The front panel, too, has more controls than one would expect to see on a "tuner only" product. Investigation of these controls, however, not only justifies their presence but tells us a thing or two about thoughtful layout and the pros and cons of consumer product miniaturization. The front panel is finished in tasteful light gold-anodized color, the tuning and dial scale area in a contrasting black, with the dial numerals themselves *almost* (but not quite) "blacked out" when power is turned off on the unit. The ends of the panel are flanked by two attractive wood blocks and, since the dimensions of the panel exceed the width and height dimensions of the chassis and metal wrap by at least a quarter of an inch all around, custom installation of the entire unit is made simple—even if your cabinet front panel has the most ragged of rectangular cut-outs. Controls along the lower section of the panel include a rotary power on/off switch, individual output level controls for AM and FM, a muting threshhold adjustment, three push-push buttons for muting on/off, MPX noise filter on/off and AFC/on-off and a three-position rotary selector switch with settings for AM, FM and

FM Automatic. A large tuning knob, located in the upper section of the panel, is coupled to an effective flywheel. Station indication is provided by means of a traveling bead of light, rather than the conventional pointer. When the tuner is tuned off station, the bead of light glows dimly, but when a station is tuned in, the bead's light intensifies. Since meters are installed in the dial scale area (a signal strength meter for both AM and FM and a center-of-chanel meter for FM), this illuminated pointer feature is somewhat redundant, but makes for an interesting conversation piece. An FM stereo indicator lamp located at the upper right of the dial scale area completes the front panel layout and features.

The rear panel layout of the Pioneer TX-900 tuner has conventional screw-terminals for a 300-ohm FM antenna connection, as well as ground and an external AM antenna. A standard

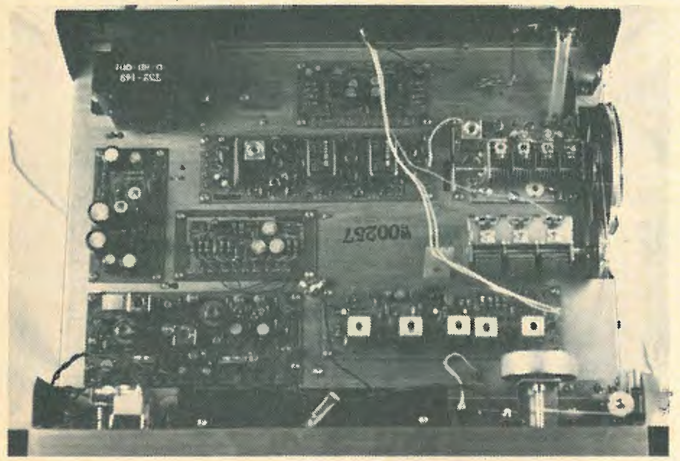
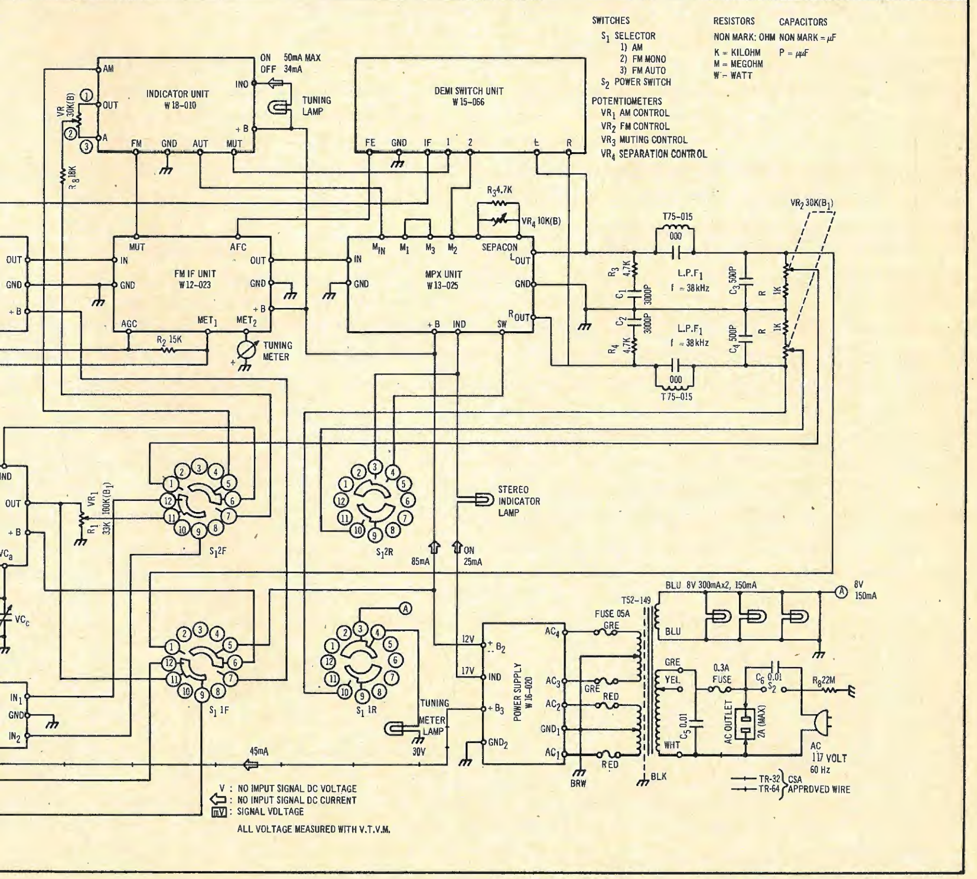


Fig. 1—Showing top view of the Pioneer TX-900

BNC coaxial connector is also provided, for connection of a 75-ohm antenna transmission line. Two sets of output jacks come next, one pair for connection to the amplifier (these outputs are controlled by the level controls on the front panel) and one pair for connection to a tape recorder. The latter pair have constant output regardless of level control settings. The AM Ferrite Loopstick is mounted on a pivoting bracket which permits the entire antenna to be swung down and away from the metal portions of the chassis for better AM reception. When we reviewed a Pioneer receiver some time ago, we criticized that company for providing a stereo FM separation control on the rear panel, maintaining that such an adjustment is strictly for factory personnel and is apt to tempt unsuspecting users into thoroughly upsetting the separation characteristics of the accurately calibrated and aligned multiplex or stereo circuitry. The TX-900 has *recessed* this separation control on the rear panel and made it a screwdriver adjustment in an attempt to discourage tampering or altering its setting. We *still* contend that this control does not belong on the outside of the chassis! Nearly everyone we know owns a screwdriver, and the temptation to rotate the adjustment is great—particularly for audiophiles. We speak from experience—we were able to reduce stereo separation from an excellent 40 dB to about 10 dB with just a minimal rotation of this control. Fortunately, we had the test equipment on hand to restore proper separation—the average users does **NOT**—and it is an adjustment that cannot be performed by listening tests. Strangely, the operating manual supplied with the TX-900 seems to agree with us. To quote it, "This control adjusts the channel separation of FM multiplex stereo broadcasts. It has already been adjusted at the factory, and normally there should be no need for any further adjustments, which are extremely critical and cannot be adjusted by ear." So why put the control on the outside, gentlemen? A fuse-holder and a convenience switched a.c. outlet complete the layout of the rear panel.

Fig. 2.—Block diagram of the Pioneer TX-900 circuitry.



Circuitry

Internal views of the chassis layout are shown in Fig. 1. Eight separate circuit modules are arranged in this spacious chassis. The front end employs a two-stage r.f. amplifier and includes three FET devices plus a bi-polar transistor in the local oscillator circuit. The all-IC i.f. section uses a combination of crystal filters and conventional interstage transformers (two stages of each) plus a variety of clamping and limiting diode circuits. The stereo FM decoder section is of the switching type and utilizes nine transistors, four of which are involved in the switching circuitry and the stereo indicator circuits. The AM section is of conventional design, using five transistors and regular interstage transformer tuning in the i.f. section. Additional modules include an indicator unit, responsible for the unique tuning lamp described earlier, an audio amplifier section (voltage amplification only), a power supply module (which includes transistorized and Zener voltage regulation of critical B+ voltages), and a small switching module containing the three push-push switching circuits described in connection with the front panel. The inter-relationship of all these sections is shown in the wiring block diagram of Fig. 2.

Electrical Measurements

Some of the more significant performance characteristics (mono FM) of the TX-900 are shown in the graphs of Fig. 3. Note, that IHF sensitivity is 1.7 μV, exactly as claimed. S/N ratio is actually better than claimed, measuring 75 dB as opposed to 70 dB claimed by the manufacturer. While THD in mono measured 0.4% (somewhat higher than the 0.3% claimed), the THD measured in stereo conformed exactly to the 0.5% claimed. Both THD figures are quite good, as a matter of fact. Full limiting occurred at less than 2 μV, just about at the IHF sensitivity input.

Capture ratio measured 1.7 dB, very close to manufacturer's specifications, whereas alternate channel selectivity exceeded claimed specs, measuring 70 dB. It should be pointed out that the output level of 1 volt (for both AM and FM) claimed by the manufacturer applies to only 30% modulation. In the case of FM, this means that at 75 kHz modulation (100%), output is about 3 volts. We actually measured 3.5 volts under these conditions. Since this is a rather high amount of output for the input of many preamplifiers and even power amplifiers, we would recommend that the output level controls be set at about mid-point to prevent overload of input stages on auxiliary equipment with which the TX-900 is to be used. The nice part about having separate volume controls for AM and FM is that each can be adjusted to equal loudness level of other sound sources connected to your preamplifier or amplifier, such as phono, tape, etc. In this way, switching from one of these sources to either AM or FM results in no drastic change of audio level.

In the course of our measurements and in our listening tests, we experimented with the three push-buttons available on the front panel. The a.f.c. button provided the usual a.f.c. pulling action and was extremely symmetrical on either side of center frequency. The MPX filter did reduce noise on weak stereo FM reception, but it could cut into the otherwise excellent separation characteristics of the tuner. For example, separation at 5 kHz was reduced from 33 dB to 22 dB when the MPX noise filter was introduced, while at 1 kHz, the 38 dB separation figure was reduced to some 32 dB. (See Fig. 4.) The mute switch, used in conjunction with the mute threshhold adjustment control was extremely effective. There was no evidence of distortion at signal strengths in the vicinity of the threshhold signal strength value. Muting can be set at anywhere from 5 μV to about 30 μV. Generally, we kept the control set for maximum sensitivity (5 μV) in our listening tests, since in no instance did this setting result in noisily received stations.

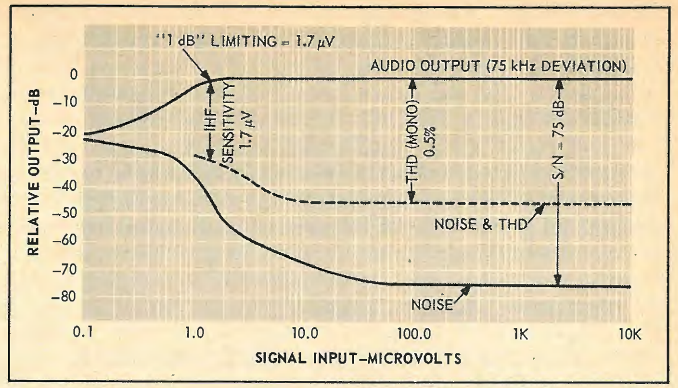


Fig. 3—FM mono performance characteristics.

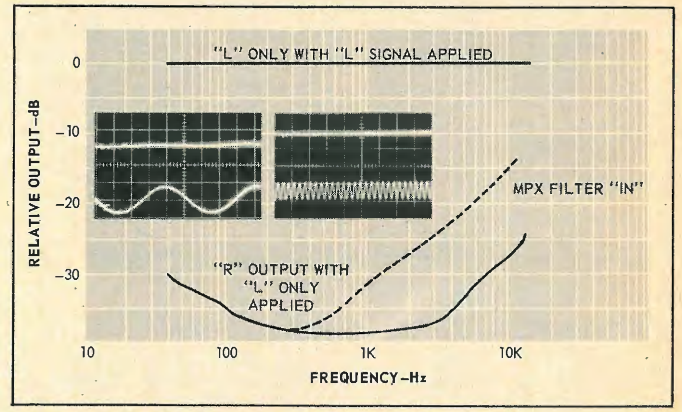


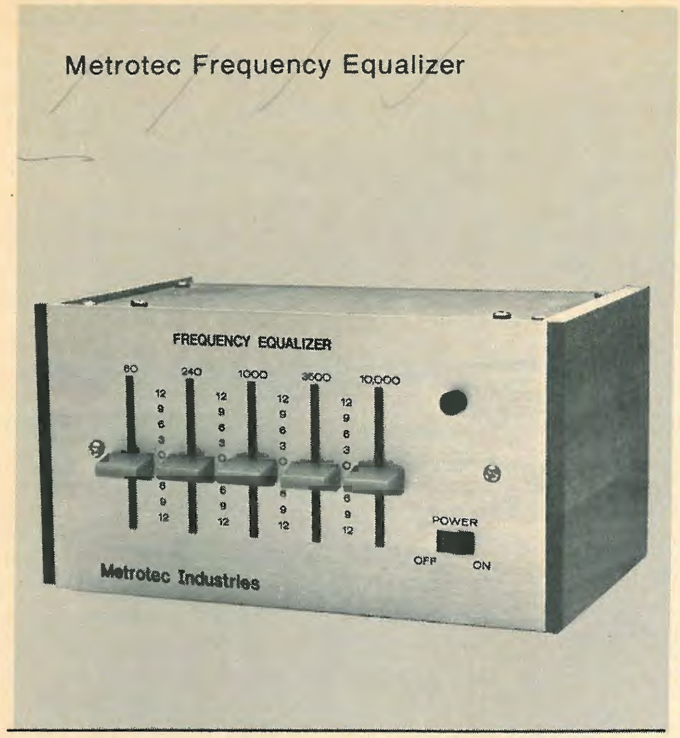
Fig. 4—FM stereo separation characteristics. Scope photo at left shows separation at 100 Hz; photo at right shows separation at 5 kHz.

Listening Tests

As mentioned in these columns before, the combination of good sensitivity *and* excellent alternate channel selectivity, plus good capture ratio, results in station selection that is smooth, devoid of any forms of interference and capable of the widest selection of acceptable station performance in this metropolitan area. We logged no less than 57 stations (using an outdoor six-element Yagi antenna and a rotator). At least 10 of these were only one channel (200 kHz) apart, and two pairs of stations were at the same frequency but originated from different points on the compass. No evidence of adjacent or alternate channel interference was noted for any of these received stations. Five of the stations listened to were engaged in supplementary SCA (background music) transmission, so we were particularly conscious of the effectiveness of the double-tuned SCA filter traps which are built into the multiplex section of this tuner. We heard no "tweets" or "swishing" sounds when listening to these SCA-operational stations, even at high volume settings and with the treble control of our associated amplifier turned fully clockwise. That is quite a severe test for SCA interference, by the way. Stereo separation was excellent and, although we were plagued with multi-path interference on approximately five of the 23 stereo stations received, that is hardly a fault of the Model TX-900 tuner.

If your purchasing trend is towards separate amplifier and tuner, or even separate preamplifier, amplifier and tuner, you would do well to consider the excellent performance and the moderate (by comparison with other first-rate tuners) price of the Pioneer Model TX-900. Surprisingly, the unit's AM (which performs quite well) is an extra bonus, since most "higher priced" tuners do not include an AM section at all. *L.F.*

Metrotec Frequency Equalizer



MANUFACTURER'S SPECIFICATIONS:
Frequency Response (flat setting): 5 Hz to 100 kHz ±1 dB. **Tone Control Range:** ±12 dB at 60.240, 1000, 3500, and 10,000 Hz. **THD:** .05% at 2-volt output. **IM Distortion:** .05% at 2-volt output. **Hum and Noise** (input shorted): 80 dB below 1 volt. **Maximum Output:** 9 volts, rms. **Gain:** Unity, ±0. -2 dB. **Recommended Output Load:** 10,000 ohms or greater. **Input Impedance:** 75k ohms. **Output Impedance:** 10 ohms. **Inputs:** Main, tape monitor. **Outputs:** Main, tape out. **Controls:** Power switch, 5 frequency controls, tape-monitor switch (rear panel). **Semi-Conductors:** 8 transistors, 4 diodes. **Power Consumption:** 3 watts, 117 volts. **Dimensions:** Height, 4¾"; width, 8½"; depth, 5½" over knobs. **Weight:** 2½ lbs. **Price:** Factory assembled, $99.95; kit, $79.95. (include $1.25 for postage and shipping if ordering direct from manufacturer, Metrotec Industries, 1405 Old Northern Rd., Roslyn, N.Y. 11579)

The great interest in selective frequency control that has followed the introduction of "Acousta-Voicing" by Altec has engendered the introduction of a number of devices which permit the user to boost or cut certain frequencies without affecting others appreciably. Among these are the Acousta-Voicette by Altec, the Advent Frequency Balance Control, and the selective equalization provided by some of the JVC receivers and more flexibly in their professional-type preamp, as well as by the Harmon-Kardon Citation Eleven.

Conventional tone controls provide boosts and cuts at both ends of the spectrum—some with variable hinge points, and some with fixed ones, the latter being the simplest and consequently the more common. But critical listeners are painfully aware that these types of controls do not always correct the perceived deficiencies in response, and most certainly they do not do much to correct acoustic conditions caused by room shapes, furnishings, and the like.

The Metrotec Frequency Equalizer offers the user the ability to vary response in five ranges spaced at about two-octave intervals, and both channels are varied simultaneously by the same amount. And this frequency equalization is done with practically unmeasurable distortion, an extremely important factor in today's installations. Better still, the completed equalizer is available at a reasonable price, with a $20 saving if you build it yourself from a kit. Kit instructions are excellent, and you should be able to complete the unit in about four hours, even with little advance experience.

The unit is equipped with the five slide controls which vary frequency over a range from + 12 dB to -12 dB at each of the frequencies. Curves for the various controls are shown in Fig. 2, with all having been measured separately then combined for the set of curves. There is some overlap of the controls, so that if you were to set the 60-Hz control to +12 and the 240-Hz control to +12, you would get a boost of +17 at 60 Hz and +17 at 240 Hz, with a dip to about +16 dB at 100 Hz, for example. Any response can be determined by adding the boosts of the controls in the boost position and subtracting the cuts of any of the controls in the cut positions. The response with all controls at the extremes of their range are shown as dashed lines above and below the individual solid-line curves.

Thus it can be seen that a great variety of overall responses can be obtained by setting the controls at whatever positions you may wish.

In fact, if all the possibilities were plotted, it would take a number of pages just for the curves alone. The important factor is that you can tailor the response to get the result you want in your particular listening room, and to a great extent you can compensate for deficiencies in the equipment. In general, all modern equipment is essentially flat over the entire audio range with the exception of the transducers. Not all phono cartridges are up to broadcast standards (though many are), and everyone knows that speakers differ appreciably. With the Frequency Equalizer, however, it is possible to compensate for differing frequency responses of speakers, and more importantly, it is possible to compensate for the room in which you do your listening. It is extremely interesting to observe the effect of reducing the 1000-Hz response by as little as 3 dB, and it is, of course, possible to tailor response to compensate for hearing itself at low levels.

The Frequency Equalizer can be connected between a separate preamplifier and power amplifier if such a system is in use. For the average receiver, however, it is necessary to connect the equalizer between the TAPE OUT and TAPE IN jacks and then set the tape-monitor switch on the receiver to the IN position, with the tape-monitor switch on the equalizer in the OUT position. When used with a three-head tape recorder, the equalizer is connected in the same manner, while for tape-playback use (or for monitoring off the source) the switch on the rear of the equalizer is set to IN.

Circuit Description

The circuit of the Metrotec Frequency Equalizer differs appreciably from that used in the JVC receivers and in the Citation Eleven, which was shown in the January, 1971 issue.

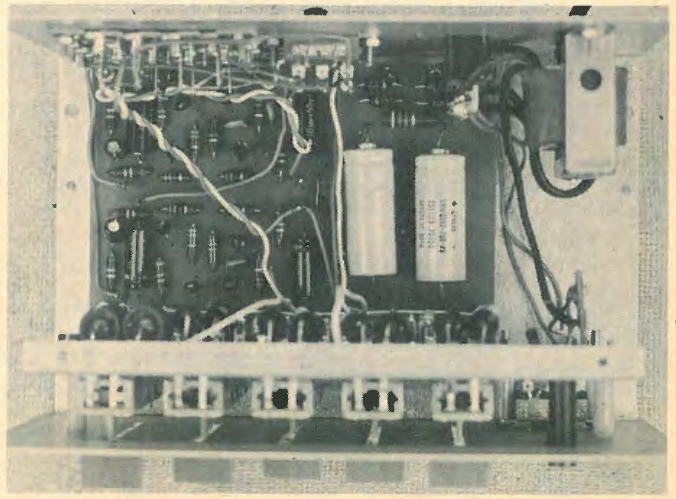


Fig. 1—Top view with cover and end panels removed.

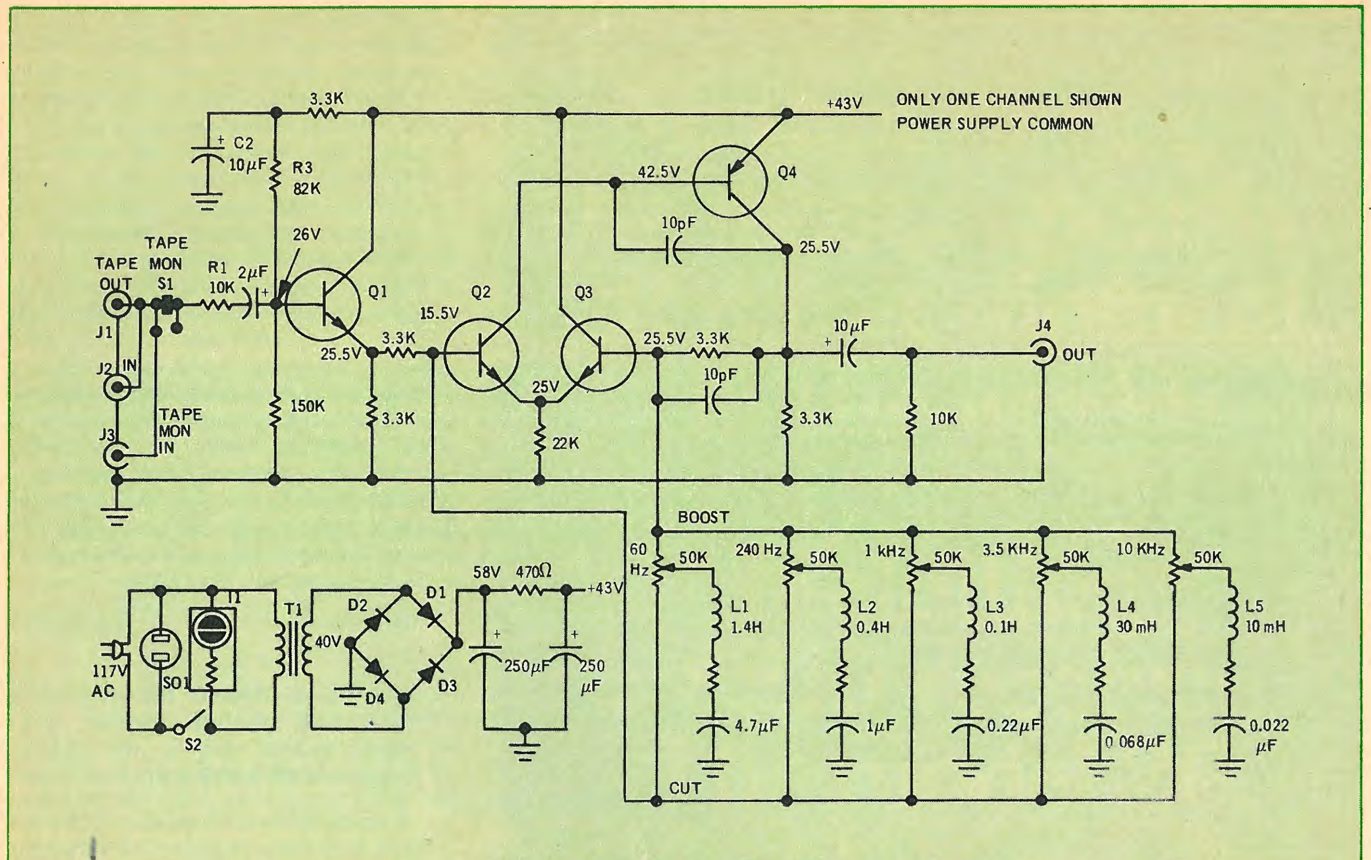


Fig. 2—Overall schematic diagram

In the Metrotec unit, the input is fed to an NPN transistor connected as an emitter follower, and its output is fed to one base of a differential pair. The collector of this transistor is fed to the output through a PNP transistor, and the output is connected back to the base of the second transistor of the differential pair through a resistor. The two bases are thus at the same d.c. potential, and connected between them are the five slide-type controls. The emitters of the differential pair are connected together and to ground through a 22-k ohm resistor, and the sliders of the five controls are connected through separate resonant circuits to ground. Thus when the slider is in the boost position, the feedback is shunted to ground at the resonant frequency, and in the cut position the signal is shunted to ground at the selected frequency. Proper choice of the various components gives the desired results. With the ends of the potentiometers connected directly to circuit points of equal potential, no coupling capacitors are required, and this results in less phase shift thoughout, and more importantly, less low-frequency attenuation. The power supply consists of a transformer delivering 40 V. a.c. to a four-diode bridge rectifier, with the d.c. output well filtered by two 250-. F capacitors and a 470 ohm dropping resistor to provide 43 V. d.c. to the circuit. The pilot light is neon, operating across the a.c. line input, and the convenience outlet is unswitched.

Performance

We found the performance of the Frequency Equalizer to conform to the specifications in every particular. Then we tried it out with a pair of high-quality speakers in our normal listening enviornment, and later in our shop area with speakers of lesser abilities. In the normal listening area it was possible to compensate for the room to a comforting degree, and especially to simulate the characteristics of the ear at low listening levels. With the 60-Hz control at maximum boost, there was still no audible hum, and it was noted that an increase of 3 to 5 dB was enought to make a great difference in organ music, for example, and with a dance band to add a sense of actually being present. Records that may have been deficient in highs on violin solos were brought up to what we thought was a more natural sound. On the whole, we are convinced that selective frequency equalization is the step of the future, even though it is here now, and for the first time, at a practicable price, particularly if you build it from a kit. *C.G. McP.*

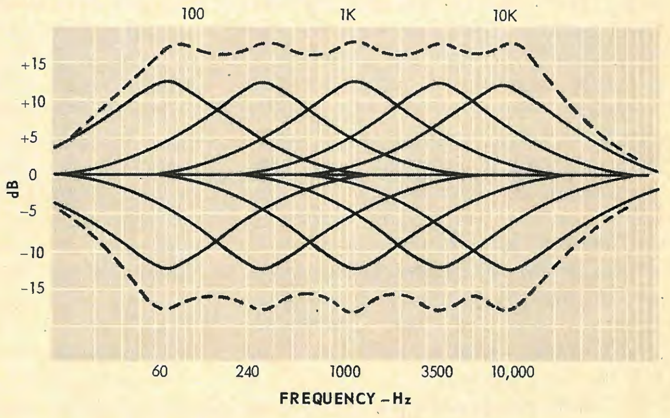
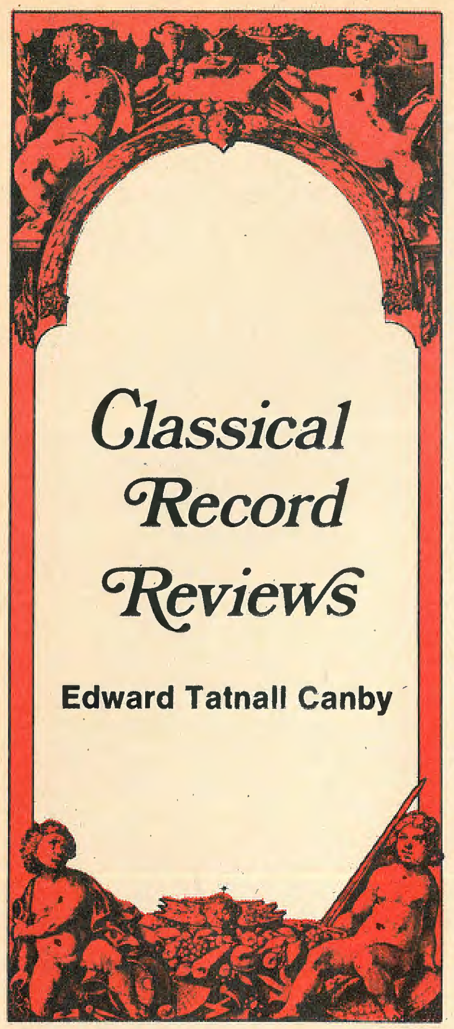


Fig. 3—Curves of responses of the five equalizer controls at maximum boost and at maximum cut positions. Dashed lines indicate response with all controls at maximum positions.



Classical Record Reviews

Edward Tatnall Canby

Berlioz: Requiem, Op. 5. London Symphony Orch. and Chorus, Wandsworth School Boys' Choir; Ronald Dowd, tenor. Philips 6700 019 (2 disks) stereo ($11.96).

The best Berlioz Requiem *ever* put onto records, I say. The best, that is, in a musical sense.

I began it with tepid interest—too many hi fi jobs in the past featuring the inevitable *Dies Irae* with its four brass bands. I ended up enthusiastic. A really wonderfully persuasive performance, getting at all the original earnestness of a youthful genius whose ideas of musical economy were nonexistant. In the wrong hands, this Requeim can be a windy bore, overblown hot air. Not here! It really justifies itself, in its own terms. Yes, there's splendid fi. But the emphasis is on good musical recording, not spectacular sound. You'll want to play it all, not just the loud parts.

Colin Davis should come first in the credits. His personal magnetism must be immense: these performers are really suffused with excitement and earnestness, both singers and orchestra. But there is no flamboyance; the musual intensity is beautifully shaped and controlled. The dominant impression is one of total *acceptance* of the music, in its own terms, a simplicity and straightforwardness that somehow minimizes the windiness of the expression and renders it sweet and clear. Good! We always knew it had this quality, but how to make it sound that way? Davis does it.

I have a dim feeling that perhaps the "London Symphony Chorus" is not always the same group. Haven't I heard it with a quite different sound? Not sure—but no matter! *This* group of ardent (mostly) young people produces a superbly blended sound, without a trace of rough vibrato to thicken the mix. It is precisely what the music needs, for purity and limpidness. Yet the sense of urgency, the straining to climaxes, the hushed, mysterious soft parts, are done with complete faith and a great deal of expertise. The boys' voices add an extra innocence that will surely remove your last doubts. This is *the* Requiem for those who love the music.

Performance: A- Sound: B+

The Triumphs of Oriana. Purcell Chorus, London Cornet & Sackbut Ensemble, Elizabethan Consort of Viols, Grayston Burgess. Argo ARG 643 stereo ($5.95)

This is pretty much of a triumph itself, of good music making an imaginative reconstruction. People who like madrigals—and people who distrust them—will be amazed. Who ever heard of madrigals done by chorus, plus orchestra of viols and brass!

The rationale is twofold. The "Triumphs," long known simply as a publication of some 25 madrigals by various composers assembled by Thomas Morley, all with the suggestive refrain, "Long Live Fair Oriana," have always been supposed to have been dedicated to Queen Elizabeth I. It seems, the new argument goes, that a similar Italian set with a similar refrain was the model—most Elizabethan madrigals were modeled on Italian originals. Moreover, a triumph was a court jousting tournament, at which there was much pageantry, usually upon some courtly theme; an old Spanish tale, Englished in late Elizabethan times and popular, had an Oriana, daughter of a British King, as its heroine. . . .So, it is deduced, these madrigals formed part of a tournament pageant known to have occured in 1592, honoring Oriana and, by implication, Elizabeth. . . .

No sooner said than done! Crowd voices begin this record, followed by heralding brass, then a noble spoken poem. Some 13 of the "Triumph" madrigals are then put on in pageant style, with astonishing pomp, making use of solo voices, choruses, accompaniment for strings, brass, with an occasional Elizabethan instrumental piece as an interlude and contrast. Some of the madrigals are almost Handelian in scope, especially in the splendid concluding "Long live fair Oriana" refrains. Great variety is worked into their texture—one is done entirely with male voices, for instance. All in all, this is a very new look at the madrigal art and, I should say a remarkably successful one in terms of musical communication.

My only reservation is in the tempi, which tend to be too uniformly the same, rather slow and a bit lumpish and marching. But after all, nobody really knows how they went.

Performance: B+ Sound: B+

John Cage—Music for Keyboard 1935-1948. Jeanne Kirsten, prepared piano, piano, toy piano. Columbia M2S 819 stereo ($11.98).

I sampled in and around these two discs with some surprise—I'd forgotten. Recent works from the mild-mannered Mr. Cage, as polite and proper a guy

as you'd ever run into in the flesh, have been something more than hair-raising as many of us have discovered. His "concept music" (music with an Idea to Propound) tends towards the screamingly raucous, whether via 13 simultaneous radios or a vast number of computer tracks or even a large batch of anquishedly overloaded loudspeakers, straining their guts in total agony. Mr. Cage's most engaging present habit is the non-stop blast. The agony is never short. It goes on, and on, until your senses shriek with the unendurable. That is precisely the point—or one of the points.

In his earlier work, challenging enough in its own way and its own day,

things are much more easily endurable. The piano pieces are often quite short and very subdued. They have even a recognizable style, vaguely French out of Satie and whatnot (note the tricky titles, like "A Valentine Out of Season," or "Prelude for Meditation"), and a fixed content, not via happenstance, calculated or otherwise—that came later. The little piece for toy piano (it had me mystified when somebody played it for me out of context) sounds like a somewhat absent-minded exercise. The prepared piano pieces, with the instrument's insides doctored up, now sound really quite—shall I say—*expected.* We hear this kind of musical sound all the time today, in concert, pop and even commercials. And thanks to Cage in the 1940s, you may be sure. It blends beautifully with much else that is now obviously becoming a familiar area of sound-structuring for all of us, even the tinniest-eared.

If you think Cage is a crackpot, listen to this, from the year 1937:

"I believe that the use of noise to make music will continue and increase until we reach a music produced through the aid of electrical instruments which will make available for musical purposes any and all sounds that can be heard."

Thirty four years ago, and right on the button. A guy who turns out prophesies like that, I suggest, should be listened to with wary respect. Right now, he is 57 and going strong. Without the slightest doubt he has a sizeable hunk of our future right in his hands, today. As for *you,* you can take it or leave it. Your risk!

Performances: Obviously competent. *Sound:* B+

Purcell: Sacred Music at the English Court. (Six anthems; Chaconne). Choir of King's College, Cambridge; Leonhardt-Consort. Telefunken SAWT 9558 stereo ($5.95).

Salisbury Cathedral Choir (Purcell, Blow, Wise, Boyce, Greene, Battishill, Croft). Dir. Christopher Dearnley. Argo ZRG 5247 stereo ($5.95)

The Telefunken offering, a collaboration between the British King's College Choir and Gustav Leonhardt's Dutch-based old-instrument ensemble, is one of the finest Purcell records I've heard. The famed British choir, never in better voice or better balance (as between the not-too-wobbly mature males and the not-too-breathy boy choristers) is born to this music and sings it earnestly, with life and dedication, along with the three

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Classical Record Reviews
(Continued from page 55)

soloists, James Bowman, Nigel Rogers, and Max van Egmond. The Leonhardt group, playing on old violins (I think restored to their earlier playing condition) plays livelier, more natural Purcell than any English group I've heard, particularly in the lively dotted tempi and the ornaments, which as a harpsichordist and organist, Leonhardt knows how to do really right. The recording was made in Holland; the Choir presumably flown over, or on tour. How splendid to hear the long-familiar "Bell Anthem," "Rejoice in the Lord Always," done with such marvelous authenticity and vigor! And the five other anthems, large and small—a segment of the 69 in the Purcell catalogue.

The Salisbury Cathedral recording is a curious one, for here we are, so to speak, in the British musical provinces, hearing a choir that is relatively isolated in its own record as compared to the cosmopolitan King's College group. Extremely British, but narrowly so. According to what seems to be a common way today, the boy choristers are thin and breathy, their voices absolutely without vibrato, whereas the older tenors and basses sport an enormous, rich tonal production, utterly at odds with those of the tiny boys' voices. These boys are *very* thin, though touchingly earnest; the men are very fat-toned and wobbly; the combination is surely arbitrary and not typical of British singing in earlier ages.

The music begins with Purcell and his older contemporary John Blow, then goes forward through a succession of minor British figures through the 18th century. Like so much admirable British church music, it is all of a piece, product of a remarkable continuity over the generations.

The choir does not come through well; the boys' voices are not only thin but somehow muffled; the men overbalance them. The soloists verge on the pretentious in their styling—except for a tiny waif of a boy soloist, whose whispy voice is barely audible; can't be more than eight or nine years old. He sings a preposterous duet with an enormous, heavy basso.

Performance: B- Sound: B-

(A) Beethoven: Complete String Quartets, Grosse Fuge. Guarneri Quartet. RCA VCS 11-100 (11 disks) stereo ($76.78).
(B) Beethoven: The Complete Quartets, Vol. 1; Op. 18. (The classic 1951 recordings). Budapest Quartet. Odyssey 32 36 0023 (3 discs) mono ($8.98).
(C) Beethoven: Quartets Op. 18, No. 2; Op. 135. Flonzaley Quartet. (Recorded 1926). Perennial 2003 mono ($6, Record Undertaker, P. O. Box 437, New York 10023).

Here is a fascinating set of comparisons—though the older Budapest set seems to have been retired, after only a year or so in the Odyssey reissue catalogue (probably through confusion with the later stereo recordings on regular Columbia) and the Flonzaley is available only via mail; whereas the RCA collection, if I am right, sells at an incredibly high price, listed as $6.98 per disc. All these recordings are assemblages from originals issued in other formats.

In my lifetime, from the early 1930s on, the Budapest was *the* string quartet, far ahead of many worthy competitors in the discipline and sheer intensity of their communication in Beethoven, Mozart, and, to a lesser extent, some of the Romantics. So intense, indeed, that their habitual tone was almost hoarse, so strenuously did their instruments produce a Twentieth Century musical voltage. That hoarseness increased over the decades. Hence the "classic 1951" recordings, already late (their earliest were, over here, on Victor 78s in the 'thirties) were worth continuing on Odyssey and hopefully will reappear. But before the Budapest, there was the legendary Flonzaley, the first serious quartet recording artists and, long before that, world famous, like the Budapest. Today, RCA is doing what it can to promote the Guararneri quartet as a successor to these, aided by much "critical acclaim" from all directions.

The three groups compare edifyingly and reveal unsuspected facets of the continuing quartet art. Imagine it—the Flonzaley, its styling formed as early as 1905; the Budapest, in its later format dating from the early 1930s, and the Guarneri, formed the day before yesterday! The recorded art is really beginning to light up the history of music, as it has never been elucidated before.

The Flonzaley recordings confirm a gradually growing thought that has been in my mind, that the further back we go towards the actual late Romantic period, the *less* "Romantic" is the actual sound of performance. I'll cite two major respects. First, tempo and shaping—earlier performances were faster, more regular in tempo and much more aware of over-all shape and drive than today's "Romantic" playings, and this in spite of quaintly antiquated rubato (hesitations), slides and the like. Second, the use of vibrato was *much* less in the early years of our century than now, both in string playing and in singing.

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Canby's Capsules...

TITLE HEAR IT THEN	CONTENT	SOUND
I Can Hear It Now/The Sixties. Fred W. Friendly, Walter Cronkite. Columbia M3X 30353 stereo ($17.98)	Columbia's distinguished series updated, Cronkite narrating. Such a proliferation of material! This runs to six sides, mostly hi fi sound; it is both moving and, often, horrifying–so near and yet (already) so far. Cronkite lends his own brand of terseness and economy.	Predominance of wide-range, undistorted voices now lends new impact to an old format, allowing longer excerpts, easier listening, a heightened immediacy, especially those now dead–Pres. Kennedy a notable example.
Remember the Golden Days of Radio, Vols. 1, 2. Narr. Jack Benny, Frank Knight. Longines Symphonette Soc. (Larchmont, N.Y. 10538) By mail.	I expected corn in these–it's there, but amazingly interesting. Brilliant idea to use Benny and Knight (the voice of the L. Symphonette)–right out of the period. Lots of nostalgic examples, including commercials. A first-rate documentary series, very communicative, if easygoing. And, in its way, important.	Was old-time radio really quite as "dead" in the sound as this? Filters doubtless used to clean up woeful old "air checks," etc.–but we did, mostly, hear nothing over 4 kHz. The sense got thru–it still does. Sounds strange, now. Our ears have changed.
Marc Blitzstein: The Cradle Will Rock. (1937). 1964 production, with piano. CRI SD 266 (2discs) stereo ($11.96)	This union-slanted prototype for later Bernstein shows–and follower of Kurt Weil's–never got staged; Bernstein finally did it in 1947, 1960, and aided in the 1964 revivals. Pretty corny as of today's "people's music" but an important score in its field, just off the edge of musical comedy.	The sound of the music is that of the 1930s– thanks to competent direction–but the audio is modern, 1964 stereo. (It started Columbia, bounced to an M-G-M release; this is a foundation-assisted reissue.)
Menotti: The Old Maid and the Thief. Margaret Baker, Judith Blegen, John Reardon, Anna Reynolds; Orch. Teatro Verdi di Trieste, Mester. Mercury SR 90521 stereo ($5.98). **Menotti: Amahl and the Night Visitors.** Orig. 1951 TV cast. RCA Victrola VIC 1512 mono ($2.98).	Once it was "Night Before Christmas"–now, every Xmas, it's "Amahl" on TV. This is the original TV cast–there's a later stereo version (diff. cast), also RCA. The simple, pious little TV opera is best-known Menotti work; "Old Mail," earlier, is similar in sound, a super-simplified Italian opera transferred to the U.S. scene, in this case for radio. (It was later staged.) "Amahl" has good 1950 mono studio sound, unbrilliant and without (later-style) reverb but entirely adequate. A bit grainy–that much more like TV, if you're nostalgic. "Old Maid" is modern stereo recording, American cast, Italian Orch. The two go together remarkably well.	
John Tavener: The Whale. London Sinfonietta and Chorus; Anna Reynolds, Raimund Herincx; Alvar Lidell, speaker, dir. David Atherton. Apple SMAS 3369 stereo ($4.98).	Wow–a "classical" record from Apple (the ex-Beatle's label). It is classical, a mod sort of opera with trick effects but also a solid score, somewhat Stravinsky-like. A brilliant job, if youthfully prolix, composed 1966 at 22; he is doing more; maybe he's a new-style bridge, between "pop" and "classical"?	The whole production is youthful, even the conductor, and longhair predominates; but the production was recorded by B.B.C. which has commissioned more Tavener. Obviously somebody to watch.
Richard Donovan: Mass; Magnificat; Antiphon and Chorale. Robt. Moevs: A Brief Mass. Kirkpatrick Chapel Choir, Rutgers Univ., Drinkwater. Charles Krigbaum, organ. CRI SD 262 stereo ($5.98).	I dunno. The trouble with contemporary music is that it spreads out over such hopelessly diffuse territory–and CRI, working for musicians, spreads our interest to the breaking point. Donovan's Yale University choral music is sturdily old-fashioned, a la early Vaughan Williams, modally dated and not very exciting. Moevs' Mass is much more contemporary, with guitar and vibraphone–I found this side worth the home listening. But will anyone like **both** sides? Maybe Yale and Rutgers people, anyhow.	
Walter Hautzig Plays Schubert. ("Grazer" Fantasy; Valses Nobles; Sonata in D.) Gemini GME 1003 stereo (McGraw-Hill Import).	The recently discovered "Grazer" Fantasy in its second recording, plus waltzes and a familiar big piano sonata. Very Central European Schubert–in spite of Curtis (Phila.) training; much Rubato, a Chopin-like approach.	Don't know this British label but it seems up to normal British pro standards, in sound and in packaging. Handsome color-photo cover.

Classical Record Reviews
(Continued from page 57)

Listen to any of the famed opera stars. Listen to the Flonzaley Quartet. An almost chaste blend of sound, with minimal vibrato in all four instruments. A businesslike pace, a long view, an immense attention to clarity of detail. Astonishing. Only an anachronistic upsliding of pitch, quite frequently, gives away the antiquity of the playing. They don't do it that way now.

In comparison, the Budapest is more intense, with a greater vibrato sound in all parts, yet still concentrated on leanness and economy of expression, for maximum shape and impact. That is as we remember this group.

As for Guarneri, a group of still-young players, the performance is most assuredly highly competent and professional, but the communication is just as clearly less, sound for sound. Compared to the other two groups, Guarneri is full of fuss, thick, overly concerned with momentary effects, very much involved with the sound of the moment and not nearly as concerned with larger shapes and over-all economy of means. Not bad–don't misunderstand. This is high competition! But Guarneri merely proves that the older groups were what they were supposed to be, each of them working for decades, a quarter century of continued association, before their respective recordings were put down.

(A). Performance: B+	Sound: B+
(B). Performance: A	Sound: B+
(C). Performance: A	Sound: C (78 electric)

(A) Beethoven: Piano Sonatas Vol. IV. Claudio Arrau. Philips PHS 3 915 (4 disks) stereo ($17.94)
(B) Beethoven: The 32 Piano Sonatas. Claude Frank. RCA Victrola VCS 9000 (12 disks) stereo. ($35.76)
(C) Beethoven: 32 Piano Sonatas (Complete Beethoven Society recordings 1932-37) Artur Schnabel. 5 vols. Seraphim ID 6063, IC 6064, IC 6065, IC 6066, IC 6067 (4,3,3,3,3, disks) mono. ($11.94, $8.96).

Artur Schnabel's 78 rpm Beethoven Society albums of the 1930s made up, when finally complete, the first total recording of the famed piano sonatas–today in their latest and most economical reissue they remain a landmark, with remarkably good piano sound (the piano recorded well in the early electric process) and, of course, a world-famous profundity of interpretation from a man who was more a thinker in musical terms than a pianist as such. No one has yet surpassed his meaningful communica-
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Weingarten Looks At
Jesus Christ Superstar

The catch-phrase, often misinterpreted, was "God is dead." The bastardizations were myriad, including the tongue-in-cheek rebuttal, "God isn't dead; he's merely unemployed."

No matter, for all seem to agree that the misunderstandings were only another symptom of the generation and communication gaps: The "dead God" was organized religion, not the spirit of a diety or energy source. And the young having shunned the affectations and ornamentations strung around the neck of original theological teachings have come full circle. "You *can* go home again," they cry, and stream headlong into religiosity.

At first it was manifested in parts of the so-called rock-pop culture via peace and brotherhood themes. But those were leading only to more fundamental searches for an absolute. And now their music unveils where the hunt truly is leading, toward a God. Gospel music surrounds the pop idiom; spirituals are revived daily; hymns are mod-ernized.

The most obvious evidence is a two-disc package that has been riding the crest of the charts for some time, **JESUS CHRIST SUPERSTAR** (Decca, DXSA 7206), a "rock opera" that tells of the final week of "the son of Nazareth" in today's terms, utilizing both today's language and its musical trends.

An ambitious project, the opera (a misnomer) is the most extensive rock composition yet, surpassing in conceptual achievement (though not in actual content) even "Hair." Though it is highly imperfect, the flaws cannot detract from the effort. And there is much of merit to be noted, by any standards.

Most amazing, when one has finished listening to the 87-minute, 16-second production, is the fact that it is the brainchild of two comparative siblings, Andrew Lloyd Webber, 22-year-old responsible for the music, and Tim Rice, 25-year-old lyricist. Webber, not incidentally, also plays piano, organ and Moog Synthesizer during the course of the opera.

Featured on the set, which includes a pamphlet that lists the performers and details the libretto, are an 85-piece symphony orchestra (conducted mainly by Alan Doggett and showcasing the strings of the City of London Ensemble), a six-piece rock combo, a jazz band (entitled, appropriately enough, Nucleus), a choir (under the direction of Geoffrey Mitchell), and a vocal group (The Trinidad Singers, led by Horace James).

Vocal soloists include Ian Gillan, of the Deep Purple outfit, who takes the title role; Murray Head, as Judas Iscariot, who, despite a weak voice, had the hit single last year of the title tune; Victor Brox, who portrays Caiaphas and who formerly appeared with Aynsley Dunbar Retaliation and is now with Lord Sutch, and Yvonne Elliman, who, despite her youth (18), has the best voice, a soft, purring melancholy-tinged thing that is perfection itself for her role as Mary Magdalene.

When it's all together, the listener finds himself immersed in the power of the presentation, unable to extricate himself even if that is desired. This occurs in spite of, or perhaps because of, the opera answering no questions about the Christ story, or theosophy, merely asking them.

There is, furthermore, an ease with which the listener can follow the legend-myth-truth, with or without libretto in hand, even though the scene changes from Bethany to Jerusalem to Pilate's house, to the Temple, to the Last Supper, to the Garden of Gethsemane, to the trial site and to the crucifixion.

And the composition's magnetism exists through each of the work's 23 cuts, even though the range of music is a blanketing of rock, pop, soul, gospel, folk, musical-comedy, light opera, and classical. It exists whether the theme is forwarded by guitar (you can pick from bass, electric, or acoustic), piano (regular or electric), organ, horns, woodwinds, percussion, or what-have-you.

The songs, it is germane, are so well integrated in the work as a whole, it is most difficult to extract them for pop singles. Out of context, much meaning is lost, at least lyrically. Only "Superstar" thus far has made an individual vocal impact, although "Overture" has been an instrumental success and "I Don't Know How to Love Him" *could* stand alone as a lovely ballad.

"Overture," naturally, foreshadows what is to come musically, but it really fails to contain the power of the entirety. In fact, despite a welcome raucousness and hectic middle segment, it mostly sounds like an opening to a Hollywood Biblical epic.

The second cut, however, sharply contrasts. For "Heaven on Their Minds" succeeds almost totally. A soliloquy by Judas, in which he warns Jesus that the people believe they've found a new Messiah "and they'll hurt You when they find they're wrong," it spotlights music that mostly is secondary to the lyrics which set the scene neatly yet acts as a driving jackhammer upon the audiophile.

Judas laments that "every word you say today gets twisted round some way," then indicates that his admiration for Jesus hasn't died but he preferred Him when He was just called a man. In addition, Judas touches on the idea of genocide, a notion relevant to today's blacks and yesterday's Indians, with an admonition:

"Listen Jesus do You care for Your race?
"Don't You see we must keep in our place?
"We are occupied—have You forgotten how put down we are?
"I am frightened by the crowd
"For we are getting much too loud
"And they'll crush us if we go too far."

An interchange between Jesus and the Apostles follows via "What's the Buzz?" —good bouncy rock, plus soul— with Mary Magdalene soothing the Man of Nazareth. He contends that "she alone has tried to give Me what I need right here and now," indicating also that it makes little sense to try to find the future in the present. "Strange Thing Mystifying," which runs together with its musical predecessor, contains the idea of Judas warning Jesus that having an associate such as the prostitute is bad for His public image. Jesus responds with the casting-the-first-stone concept.

"Everything's Alright" is next, with its breezy pop-jazz and soft ballad exchanges. Jesus here predicts trouble ahead for Himself, an inkling furthered by the lyrics of "This Jesus Must Die," the following cut, in which the Pharisees and priests claim He is a threat to them and demand "for the sake of the nation" He be killed. All the while, the mob outside shouts adulatory phrases.

"Hosanna" begins the second side, with Jesus telling the crowds to leave. A Latin tinge in the background, aided by choral effect, is still another musical contrast.

"Simon Zealotes," a soul-gospel-rock insert, finds Johnny Gustafson in fine voice as Simon asking Jesus to "add a touch of hate at Rome" so He can rise to a greater power. This is coupled with "Poor Jerusalem," a slow, bluesy (and, for the most part musically, ineffectual) offering in which Jesus "puts it all together."

Jesus explains that no one else understands what's happening—but only because they choose not to. "To conquer death you only have to die, you only have to die," He chants.

"Pilate's Dream," which features the voice of Barry Dennen as Pontius Pilate, forecasts, in the folk-music idiom, martyrdom for Jesus, blame for the Roman leader. Then, in "The Temple," containing operatic counterpoint, screaming soul, and rock, Jesus ousts the moneylenders and merchants who, in His judgment, have turned a house of prayer into a "den of thieves." He then tells the crowd, which has been imploring His help with their afflictions, to "heal yourselves."

A reprise of "Everything's Alright," now the epitome of ironic comment, is followed by Mary's soliloquy, "I Don't Know How to Love Him," in which the strings lend an air of reflection to her telling of an inner change and her inability to cope with her feelings toward Jesus.

"Damned for All Time," with a heavy electric sound that leads to the rock and roll effect of the early '50s (replete with boogie and soul), and "Blood Money" are coupled. Judas, of course, at this point sells out the carpenter, telling the Establishment where He can be found when crowds do not surround Him.

"The Last Supper," the longest cut (7:10), begins the third side. Classical in orientation, but with hard rock meshing midstream with an operatic effect, it sing-songs the chorus of Apostles contemplating their own immortality through association with Jesus. He, in turn, tells them the end is near and predicts betrayal and denial.

"Gethsemane (I Only Want to Say)," a hard blues entry, is a soliloquy by Jesus in which He shows doubts of His own position.

"When we started," He sings, "then I was inspired; Now I'm sad and tired."

He questions whether it's all worth it, asking God for confirmation, and finally accepting what He must do.

"The Arrest," with its elements of Gregorian chants, church masses and rock; "Peter's Denial," the shortest track (1:27) except for the "Everything's Alright" reprise, and "Pilate and Christ," reminiscent of musical-comedy interaction, continue the story as Pilate shifts to case to Herod.

"King Herod's Song," still another musical soliloquy, is perhaps the oddest composition of all, with its emphasis on old-time rinky-tink blues, ragtime. As for the story line, it is here that Jesus is asked to perform a miracle.

The final side begins with "Judas' Death," featuring hard rock and squealing voice (supposedly to mirror agony), a church-like interlude, and a melancholy lament about his deed. Judas blames Jesus for "using" him, claiming Jesus is the murderer.

"Trial Before Pilate (Including the 39 Lashes)" is almost a copy of Gilbert & Sullivan operetta, strangely out of place considering what is happening plot-wise.

But "Superstar," the hit that preceded the album, follows with its rock-soul and hard-hitting lyrics summing up what it's all about. Says the Chorus:

"Jesus Christ, Jesus Christ
"Who are you?
"What have You sacrificed?
"Jesus Christ Superstar
"Do You think You're what they say You are?"

And one stanza says, via the voice of Judas:

"Tell me what You think about Your friends at the top
"Who d'You think besides Yourself's the pick of the crop?
"Buddah, was He where it's at?
Is He where You are?
"Could Mahomet move a mountain or was that just PR?
"Did You mean to die like that?
Was that a mistake or
"Did You know Your messy death would be a record-breaker?
"Don't You get me wrong—I only want to know."

The last two musical entries, "The Crucifixion," with its mystical, eerie tones and avant-garde jazz, plus wailing chorus, and "John Nineteen: Forty-One," an instrumental with mournful violins, heavily leaning on classical motifs, seem anticlimactical. The calm after, instead of before, the storm.

But whatever your personal choice of gods, whatever your aims in life (this, or, if you believe, the next), whatever your taste in music, "Jesus Christ Superstar" must be heard. Simultaneously, it's a happening, a thought-provoking work of art, a forward thrust in musical acumen. "Tommy," The Who's rock opera and the first attempt at the genre, pales by comparison.

This one, despite its failings, is a piece of tomorrow today.

Classical Record Reviews
(Continued from page 59)

tion of the late sonatas, those half-crazy piano works (just look at them on the printed page!) which have turned out to be the most sublime expressions of Beethoven's own thinking. In spite of profundity, the essential warmth of Schnabel's personality got through and still can impress us.

Claude Frank, of the middle-young generation, was one of Schnabel's last students, German born but a New Yorker for many years. His, too, is an amiable and pleasing personality (for years he was the accompanist and associate conductor of the New York Dessoff Choirs, where I knew him) and of all current recordings I find his versions the most immediately accessible—particularly for the aspiring beginner. Frank's approach is somehow unassuming, friendly. No furrowed brows, no heroics! The very sound of his playing instils confidence—this, he seems to say, is *music*, so why be afraid? Along with this directness, however, there is plenty of power and a wealth of casual but beautifully worked out detail. Not surprisingly, the Frank readings are at their best in the early sonatas, to which he gives a great deal more than the somewhat precious attention many of the older pianists deign to lavish on them. Frank takes them as big works and this, in its way, is an important and contemporary revelation. The late-late sonatas are technically proficient but, I think, lack that ecstasy, that grand cosmic pianist's view, that seems to come only from the vantage point of deep maturity.

The Frank recordings are all in a single volume of twelve new-style RCA thin Dynaflex discs. I'd call it a superb bargain at the low Victrola price.

Volume 4 of the continuing Arrau series is less easy to evaluate. Arrau is undoubtedly one of the big pianists of the elder generation and a representative of the small group of older pianists who can and do tackle this enormous body of music as a whole, a "package." His music has always rubbed me the wrong way and I can only say that his Beethoven rubs me less the wrong way than I expected. Yes—I know. The man is "acclaimed" the world over and all that. Nevertheless, I have found his playing cold and unmusical and I still find it so—if one assumes a very exalted plane of comparison in this case. Not at all easy to pin down to details; I can only mention, on the one hand, a vague lack of "soul"—whatever that is—and, on another, what seems to me to be unmusical phrasing and shaping, blurry pedaling. But can it be proved? No, as Philips will immediately tell me, and rightly. For *your* ear (if you trust your ear), he may well be a saint of musical saints.

Perhaps the best summary of the Arrau viewpoint is, oddly, contained in the spoken LP side which accompanies this Vol. IV (for free). It is a long monologue by Arrau on the meaning of Beethoven and—again for my ear—it places Arrau exactly. Born Chilean, he spent long years in Berlin and his account of the Beethoven rationale is 110 percent Teutonic, straight out of the late Nineteenth century. It has been a long time since I have heard such an *echt*-Germanic recital of super-hero stuff, nor such a methodically faithfull rendering of *clichés* that were already out of touch with the modern age in 1900! No disrespect intended to Beethoven; but one must somehow indicate in one's language that Beethoven is relevant to a continuing *present*. Also, I might say, in one's playing.

(A) Performance: B	Sound: B+
(B) Performance: B+	Sound: B+
(C) Performance: A-	Sound: B-

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Performance: B+	Sound: B+

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Classified

(Continued from page 65)

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WANT TO KNOW MORE about that Hi-Fi component you're thinking of buying? Would you like to get in touch with people who know about it first-hand? For more information please send your name and address to INEX, 1232 Wilson, Palo Alto, Calif. 94301. Please include 25¢ to defray processing and mailing expenses.

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Advertising Index

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APRIL 1971 ISSUE

THIS CARD EXPIRES JUNE 30, 1971

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3 ☐ Professional Recordist
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2			17			32			47			62			77			92			107			122		
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4			19			34			49			64			79			94			109			124		
5			20			35			50			65			80			95			110			125		
6			21			36			51			66			81			96			111			126		
7			22			37			52			67			82			97			112			127		
8			23			38			53			68			83			98			113			128		
9			24			39			54			69			84			99			114			129		
10			25			40			55			70			85			100			115			130		
11			26			41			56			71			86			101			116			131		
12			27			42			57			72			87			102			117			132		
13			28			43			58			73			88			103			118			133		
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APRIL 1971 60c

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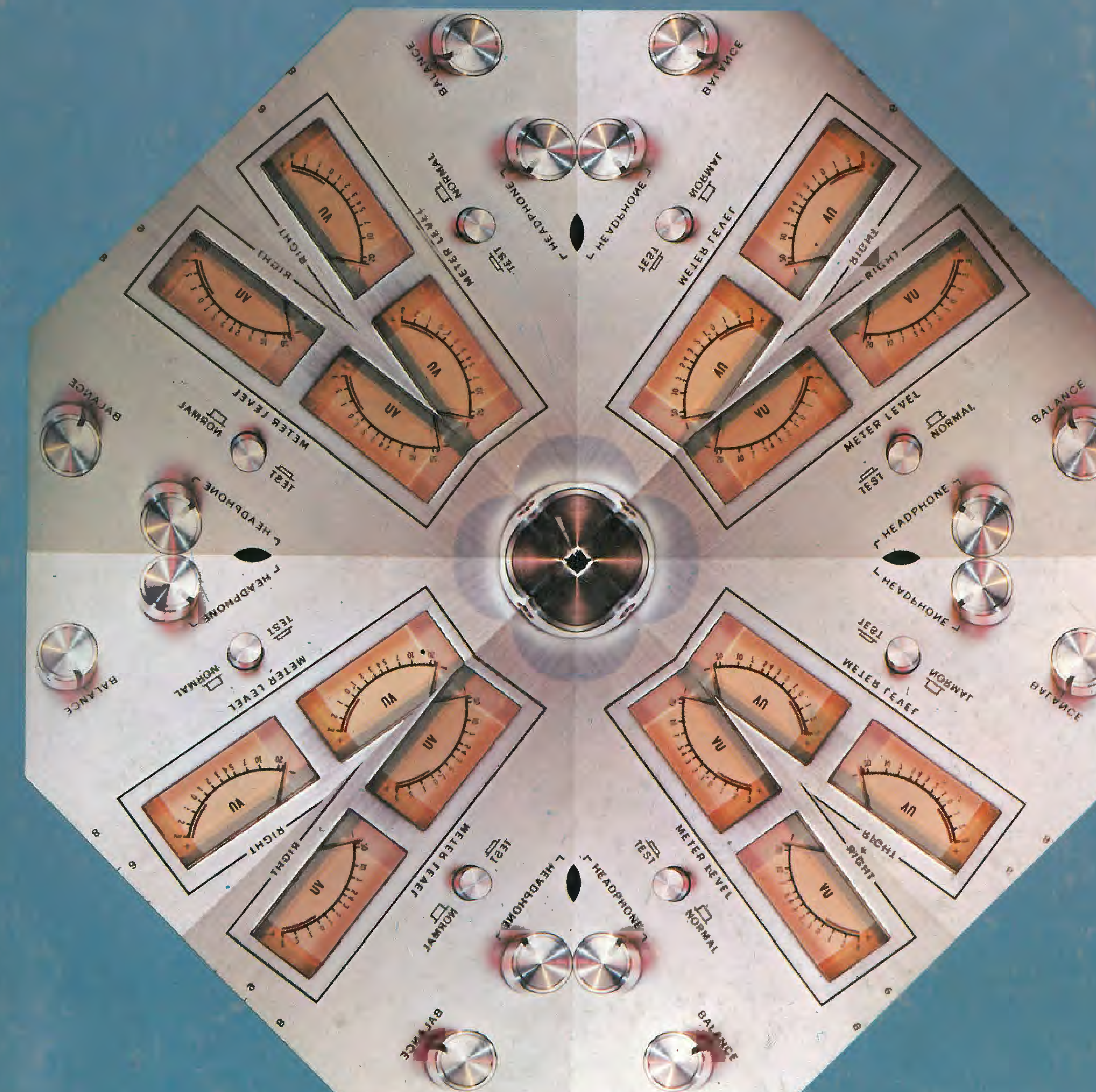

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