Heyser On Phase

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47425

Mikes Of The Pre-War Era

Mike Technique And Sound Effects

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t of each is up to us.

professional 10½-inch tape reels. Its unique combination of bias and equalization switching controls give 12 different settings to optimize the performance of any tape on the market.

The RT-1050's 3-motor transport system is activated electronically by full logic, solid state circuitry, triggered by feather touch pushbutton controls. Its transport is completely jam- and spill-proof, permitting you to switch from Fast Forward to Fast Rewind, bypassing the Stop button.

The RT-1050 was specifically designed for easy operation with a wide combination of professional features like extended linearity VU meters with adjustable sensitivity, mic/line mixing, pushbutton speed selection and reel tension adjustment buttons. There's also an exclusively designed pause control and independent control of left and right recording tracks.

The same 2-track recording system studios use for better signal-to-noise ratios and higher dynamic range is incorporated into the RT-1050. Yet it can be easily converted to 4-track use with an optional plug-in head assembly. Everything considered it's the most versatile open-reel deck you can buy. Professionals prefer it for its studio-quality performance. Everyone appreciates its completely simple operation.

Pioneer open-reel and cassette decks are built with the same outstanding quality, precision and performance of all Pioneer stereo components. That's why, whichever you choose, you know it's completely professional and indisputably the finest value ever in a studio-quality tape deck.

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U.S. Pioneer Electronics Corp., 75 Oxford Drive, Moonachie, New Jersey 07074
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Giving you the best

High fidelity is important to us at Pioneer. It's all we do and it's all we care about. We are excited that cassette tape decks have reached a level of performance that meet the highest standards. We are excited because we know that it means more enjoyment for you from your high fidelity system. We also know that you can now get more versatility and more value out of your high fidelity system than ever before.

The great advances in cassette technology have had impact on the reel-to-reel tape deck concept as well. We believe that the era of the small, inexpensive 7-inch reel tape deck is past. Neither its convenience nor its performance make it a good value compared to the new cassette technology. And it is now possible for Pioneer to offer you a professional studio-quality 10½-inch reel deck at prices that compare favorably with what you might expect from old fashioned 7-inch reel units. In our judgment the old ideas must move aside for the new ideas. And Pioneer has some very intelligent new ideas in tape for you.

The convenience of cassette. The performance of open-reel.

The stereo cassette deck has become a "must" in complete high fidelity systems. Because of its convenience, price and performance, it has virtually replaced the once popular 7-inch open-reel deck. As Julian D. Hirsch, prominent audio reviewer put it, "The best cassette machines compare favorably with a good open-reel recorder in listening quality." Pioneer proves it with four top-performing models.

Stacks compatibly with other components.

Our new CT-7171, with built-in Dolby, is a deck with a difference. It's designed with all controls up front so you can stack other components on or under it. Even the illuminated cassette compartment is front loading, for easy access and visibility.

Performance features stack up, too. Bias and equalization switches provide optimum recording and playback for every type of cassette tape made. You'll produce distortion-free recordings consistently with two oversized, illuminated VU meters plus an instant-acting peak level indicator light. And for those unpredictable program source peaks, there's a selectable Level Limiter circuit. It's similar to the type used in professional recording studios to prevent "clipping" distortion.

Finding a desired program point in a recorded cassette is simple with our new CT-7171. A memory rewind switch, working together with the 3-digit tape counter, plus an exclusive Skip button, lets you monitor audibly at accelerated speed to make precision cueing a breeze.

Automatic tape-end stop, dual concentric level controls, separate mic/line inputs, pause control, in addition to many other features, make the CT-7171 the recording studio that fits on a shelf.

Whether you choose the sophistication of the CT-7171 or Pioneer's CT-5151, CT-4141A or CT-3131A, which share many of its features, you're assured optimum performance and maximum value. One tradition that never changes at Pioneer.

Open-reel. A professional recording studio in your home.

Professionalism comes with all three studio-quality open-reel models. The RT-1020L (7½, 3¾ ips) is unequalled in 4-track units. With three motors and three heads, it has virtually every professional feature you'd want. Yet it's extremely simple to use. In addition to stereo record/playback, it also highlights 4-channel playback. The complete extent of its capabilities becomes apparent only after you've worked with it. Then you'll recognize the magnitude of Pioneer's accomplishment.

Our RT-1050 is a 2-track, 2-speed (15, 7½ ips) 3-head deck which, like all our open-reel models, can handle

12 Bias & Equalization settings optimize performance.

Whether you use a cassette or open reel deck is up to you.

Pick The Open-Reel Deck Features You Need

Model	RT-1050	RT-1020H	RT-1020L
Maximum Reel Size	10½"	10½"	10½"
Speeds	15 & 7½ ips	15 & 7½ ips	7½ & 3¾ ips
Number of Tracks	2 (4 optional)	4	4
Wow & Flutter (at high speed)	0.06%	0.06%	0.10%
Frequency Response* (±3dB)	30Hz-22kHz	30Hz-22kHz	40Hz-20kHz
Tape Bias Selection	3 position	3 position	3 position
S/N Ratio	57dB	55dB	55dB
Equalizer Selection	4-Position	2-Position	2-Position
Mic/Line Mixing	yes	yes	yes
LED Peak Indicator	yes	no	no
Memory Recording	yes	yes	yes
VU Meter Scale Selection	yes	no	no
4-Channel Playback	no	yes	yes
Motors	3	3	3
Price	$699.95	$649.95	$649.95

Pick the Cassette Features You Need

Model	CT-7171	CT-5151	CT-4141A	CT-3131A*
Dolby Noise Reduction	yes	yes	yes	no
Tape Selection	Bias & Equal.	Bias & Equal.	Bias & Equal.	Equalization
Auto. Tape Stop	yes	yes	yes	yes
Memory Rewind	yes	yes	no	no
Pause Control	yes	yes	yes	yes
Freq. Response*	30-16,000 Hz	30-16,000 Hz	30-15,000 Hz	30-15,000 Hz (*Chrome Tape)
Peak Indicator	yes	yes	no	no
Level limiter	yes	yes	no	no
Skip cueing	yes	yes	yes	no
Signal/Noise (Dolby)	58 dB	58 dB	58 dB	—
S/N (Less Dolby)	48 dB	48 dB	48 dB	47 dB
Tape Heads	Ferrite	Ferrite	Permalloy	Permalloy
Motor Type	DC Servo	DC Servo	DC Servo	DC Servo
Wow & Flutter (WRMS)	0.10%	0.12%	0.13%	0.13%
Price	$369.95	$269.95	$239.95	$179.95

*not shown

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Audio

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AUDIO (title registered U.S. Pat. Off.) is published monthly by North American Publishing Co., Irvin J. Borowsky, President; Frank Nemeyer, and Jay L. Butler, Vice Presidents; R. Kenneth Baxter, Vice President/Production: Nate Rosenblatt, Promotion Director; Mary Claffey, Circulation Director.

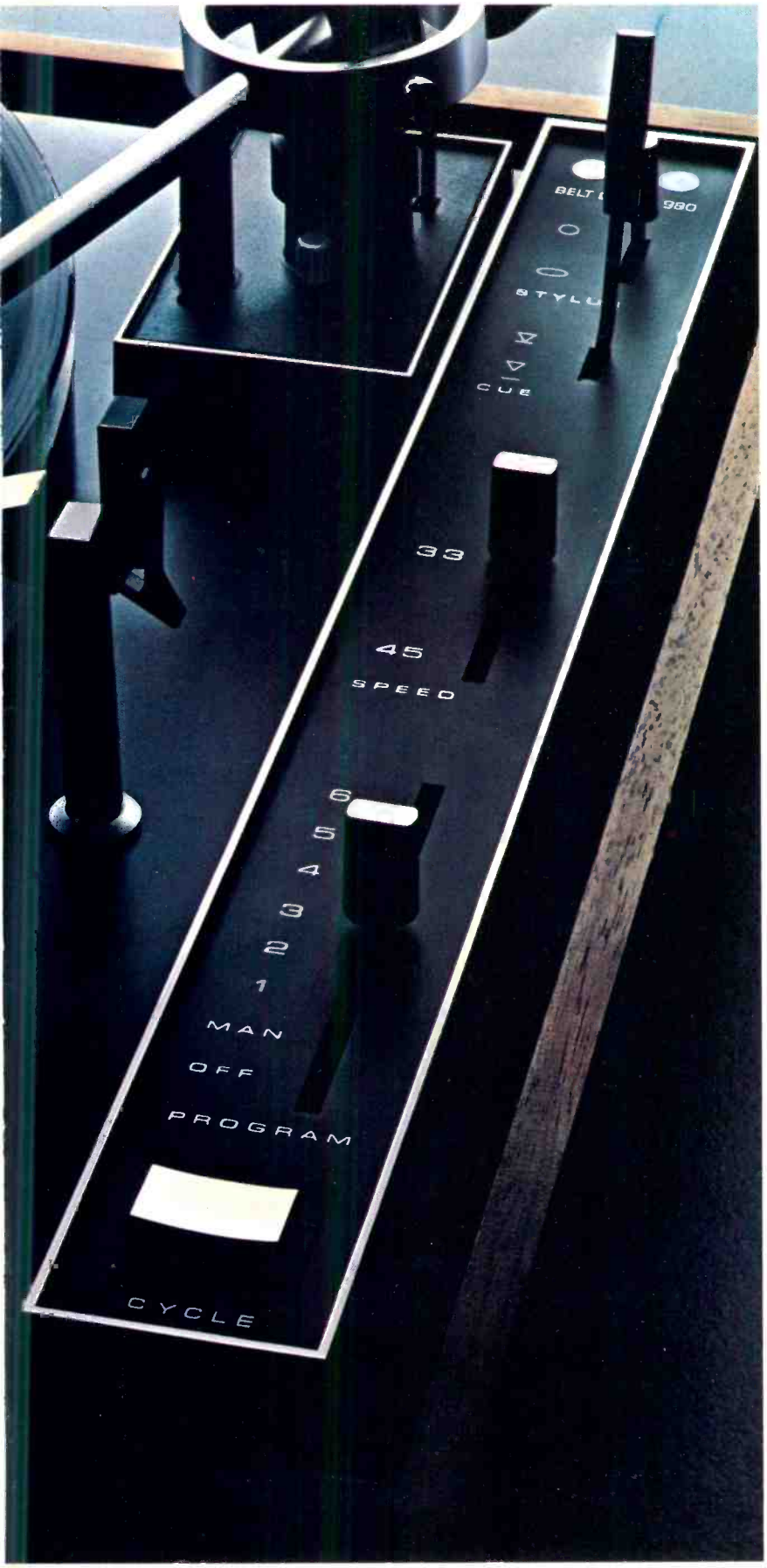
RATES—United States only: 1 year for $7.00, 2 years for $12.00, 3 years for $17.00; outside the U.S.: 1 year for $9.00, 2 years for $16.00, and 3 years for $23.00. Printed in U.S.A. at Columbus, Ohio. All rights reserved. Entire contents copyrighted 1974 by North American Publishing Co. Second class postage paid at Philadelphia, Pa. and additional mailing office. Back issues, $2.00 each.

REGIONAL SALES OFFICES: Jay L. Butler, Publisher and Sanford L. Cahn, Marketing Director, 41 East 42nd St., New York, N.Y. 10017, telephone (212) 687-8924.
Jay Martin, 15010 Ventura Blvd., Sherman Oaks, California. 91403, telephone (213) 981-7852.

REPRESENTATIVES: Continental Europe: John Ashcraft, 12 Bear St., Leicester-Square, London W.C.2, telephone 930-0525.
For Benelux and Germany. W.J.M. Saunders, Mgr., Herengracht 365. Amsterdam, Holland, telephone 24.09.08.
Japan: Japan Printing News Co., Ltd., No. 13.2 Chome Ginza Higasi, Chuo-ku, Tokyo, telephone 541-5795.

AUDIO Editorial and Publishing Offices, 134 N. 13th St., Philadelphia, Penna. 19107
Postmaster: Send Form 3579 to the above address

THIS IS A PROGRAM PANEL. WITH IT YOU CAN MAKE B·I·C™ PROGRAMMED TURNTABLES DO THINGS NO TURNTABLE HAS EVER DONE BEFORE.



The B·I·C 980 and 960, like many fine turntables, use a belt drive system.

What's unusual, however, is that B·I·C turntables can be programmed to play a single side as many as 6 times... or to play as many as 6 records in series. 'Til now, no belt-drive turntable has been able to do that.

How it works

The program lever (second from the bottom in the picture at left) gives you 22 possible ways to play your records in manual and automatic modes.

By moving the lever to "MAN", the turntable is turned on and can be operated as a manual unit.

By moving the lever to "1", and tapping the cycle button lightly, one record can be played fully automatically.

By moving the lever to 2, 3, 4, 5 or 6, you can play a single record 2, 3, 4, 5 or 6 times.

And this same program lever controls multiple play. If, for example, you want to play 2 records, simply put them on the spindle and move the lever to "2". Or move the lever to "3" and the second record will repeat once. Or move it to "4" and the second record will repeat twice. And so on, and so forth.

Must be seen to be appreciated

This program system is news all by itself. But it's far from the whole story.

The B·I·C tone arm has features found on no other tone arm.

The B·I·C motor is a major improvement over motors in other belt and idler drive turntables.

But features aside, what's truly worth close scrutiny is how all these new ideas are welded into a perfectly balanced system which performs impeccably.

We'll send you more information about the 980 and 960 if you write to:

Andrew Stephens, Dept. 12A,
British Industries Co., Westbury, L.I. 11590.

But you really must examine them, touch them and compare them, to appreciate their fundamental excellence. After you've looked them over at your B·I·C dealer's (the leading audio specialist in your area) we think you'll be impressed.

B·I·C™

This is the 980 with solid state speed control and strobe. About $200.
The 960 is identical except for these two features. About $150.

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You want pure, unadulterated music. You're entitled to pure, unadulterated music. What you get, frequently, is something less than that . . . less than you paid for. Unless of course, you already have a **Crown DC-300A Amplifier.**

The quality of reproduction from your present music library need not be strained, clipped, distorted or muffled. Nothing should come between you and the original music . . . its tones, its textures, its nuances.

"Like lifting a curtain," was how one **Crown** owner described the experience.

That's what the **Crown DC-300A** is all about. Hear it, you'll believe it.

Make this simple comparison:
(1) listen with a critical ear to your favorite recording at home; then
(2) listen to that same recording with the **DC-300A** at your **Crown** dealers. That's all.

We rest our case on your ears.

Is Crown crazy?

To guarantee **parts** and **labor**, and pay for **round-trip shipping** for three full years. (We'll even send you a shipping carton if you didn't save yours.) That takes nerve . . . and faith in your product!

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WHEN LISTENING BECOMES AN ART.

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Audioclinic

Joseph Giovanelli

Speaker Impedance Ratings

Q. I own a pair of JansZen Z-600 speakers which I drive with a Dynaco PAS3 and Stereo 70 combination. This system has been just fine during the entire eight years that I have had the system.

I met another owner of the identical speaker model at a recent party. He left me with a vexing problem; it is more in my head than in my ears (though there is a connection, yes?).

To be brief, he flatly stated that, despite the manufacturer's claims, the Z-600 is not an 8-ohm speaker. My friend measured it as having a 4-ohm impedance. (He made this measurement with his ohmmeter.) Needless to say, I was startled by this information. On the following day, I borrowed an ohmmeter from a neighbor. I disconnected the speakers and measured them. Much to my surprise and chagrin, the speakers were 4.8 and 4.6 ohms respectively.

Is there something else that I am missing? I have no knowledge in the technical area of audio, so I am sure there must be a simple explanation here. I really cannot believe that the manufacturer would deceive the consumer in this way. It just does not make sense.

If it were so, is there any possibility of damage to either of the speakers or to the amplifier because of an incorrect output connection?—David Kraft, Jackson Hts., NY

A. The impedance of a speaker is not measured with an ohmmeter. You can measure d.c. resistance with an ohmmeter, however, and this is what you have done. D.c. resistance, however, is not the same as impedance which takes into account inductance as well as d.c. resistance. (Inductance is present where a.c. is involved, and tends to oppose the flow of such current, just as resistance opposes the flow of current, both d.c. and a.c.) Unless you have access to a variable resistor, an audio generator, and a meter capable of properly reading a.c. voltage, in addition to the ohmmeter, you cannot measure impedance under home conditions.

Even if your speakers were four ohms rather than eight, no harm could come to your amplifier, even with wrong connections. Perhaps you would obtain somewhat less power than you would if the proper impedance tap was selected, but this would only mean that your speakers would be driven less hard than otherwise. Therefore, no damage could occur to the speakers.

While it is sometimes impressive for an amplifier to have the highest possible numbers related to power output, there is no advantage at all for a manufacturer to mislead the consumer by a false indication of his speaker's impedance. I think we can assume that the speakers are properly manufactured at their rated impedance.

Cueing with Headphones

Q. If I am playing one phonograph, how can I listen on a set of headphones or whatever, so that I can cue another record on the idle turntable and without the sound of the cueing coming through the main loudspeakers?—James Golden, West Roxbury, MA

A. The problem of cueing with headphones can be solved by the use of one of the small, headphone amplifiers which are readily available. Means must be provided so that it can be switched between one and the other of your phono equalizer outputs. In other words, you must have a means for paralleling the mixer and the equalizer with the headphone amplifier. The switch will connect the headphone to either of the pre-amplifier/mixer connections, but not both at one time. When a disc is playing for the dancers, you would turn down the other mixer controls associated with the second table. You could then cue up the disc on that table without its being heard in the room.

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.

The silence is deafening

The new BOSE 4401 . . . it just may be the quietest stereo preamplifier available today. Its unique MET™ (Minimum Effective Temperature) phono circuitry represents a breakthrough in low-noise technology. And the BOSE 4401 is four-channel discrete, with optional SQ and CD-4 decoding modules.* Its incredibly flexible taping facilities permit full use of tone controls and filters in both the record and playback modes. These and many other advanced features make the BOSE 4401 preamplifier an important addition to your home stereo or quadriphonic music system.

The introductory price remains in effect through December 31, 1974. See it now at your authorized BOSE dealer.

A 16-page comprehensive guide on the 4401 is available for $1. Write BOSE Corporation, Dept. AF, The Mountain, Framingham, MA 01701.

BOSE 4401

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There are 159,090 reasons to buy it.

159,090 owners had their reasons for choosing a TEAC 1200/2300 series over any other tape deck. Universities, record companies, audiophiles, musicians, audio testing labs and recording studios – all with different reasons.

Yet all with the same reason: Reliability.

The innovation behind this unparalleled performance record? TEAC's 3-motor/3-head tape transport system. (Three heads for the individual functions of erase, record and playback. And three motors, driving feed and takeup reels, and the capstan.) Our 1230 became the yardstick of the industry.

Our 2300S is the same but better, with significant electronic improvements.

Total touch-button control with logic circuitry now enables you to shift instantly from fast forward to fast rewind, and to record from pause or directly from playback. With total remote capability. Bias and EQ switches adjust for the new tapes. And there's more.

Check out the classic 2300S. And if you decide that it is unbeatable in performance and price, just remember – you're not alone.

The classic 2300S.

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Behind The Scenes

Bert Whyte

THE THIRD act of the Audio Engineering Society road show, otherwise known as the 49th Convention, opened on September 9th in its usual venue, the Waldorf-Astoria in New York.

As in past years, the ballroom of the Waldorf was crammed with exhibits of the sleek, glittering equipment of professional audio engineering. As usual, manufacturers who offer active demonstrations of their products were ensconced in rooms on the fifth floor. The papers presented at the various sessions were as informative and interesting as ever and, for the most part, well attended. New at this convention was a series of four seminars: "Introduction to Programming for Desk-top Computers," "Application of the Desk-Top Computer to Audio Engineering," "Tape Recording Alignment—Why, What, and Where," and "Practical Studio Acoustics."

In other words, the usual ingredients for a successful AES convention were on hand. Yet I couldn't help feeling that the show was a bit subdued . . . rather low key. Maybe it was a reflection of the depressing times in which we are living. Perhaps it was due to the fact that there wasn't as much new equipment nor new ideas, as we have come to expect at the AES conventions. In any case, whatever it is that gives a show "pizazz" was a bit diluted.

Which is not to say that there weren't some highlights, and some very interesting items. It would be a very rare AES convention indeed that was totally bereft of provocative ideas and innovative equipment! In my report on the 48th AES convention, I singled out the new BASF "Unisette" ¼-in. tape cassette and Howard Holzer's digital cutting-lathe control system as very significant advances in audio technology. Those who had somehow missed the introduction of the "Unisette" concept in Los Angeles were given the full run-down on the system in a paper presented by Klaus Goetz of BASF. It appears that Willi Studer will still produce the first recorders that utilize the "Unisette," however, I have been told that the introduction of these units will be somewhat later than anticipated, with a good guess being late Spring of next year. It would also appear that the original cost of the recorders, stated as being "around 400 dollars" was *extremely* optimistic. I could not elicit any further information as to the "Unisette" activities of Sony or Teac. Nothing really surprising about all this . . . "start-up time" is usually a problem with most new audio products. With an idea as exciting as the "Unisette," you can be sure I'm going to watch its development very closely.

At this convention Howard Holzer was to have presented a paper on his "Advance Headless Variable-Pitch/Variable-Depth Lathe Control System." I am very sorry to report that Howard was killed in a private plane crash in Mexico, several weeks before the convention. This tragic accident has deprived the industry of a most dynamic man with one of the keenest minds in disc recording technology. Howard's son Mark presented his paper, and I understand will continue the operation of Haeco. Although I am not certain of all the facts, I believe that Atlantic Records, here in New York, has, or will shortly install, the first of the Haeco (Howard Holzer's company) digital-delay cutting systems. I look forward to hearing some Atlantic discs cut with this clever, new system.

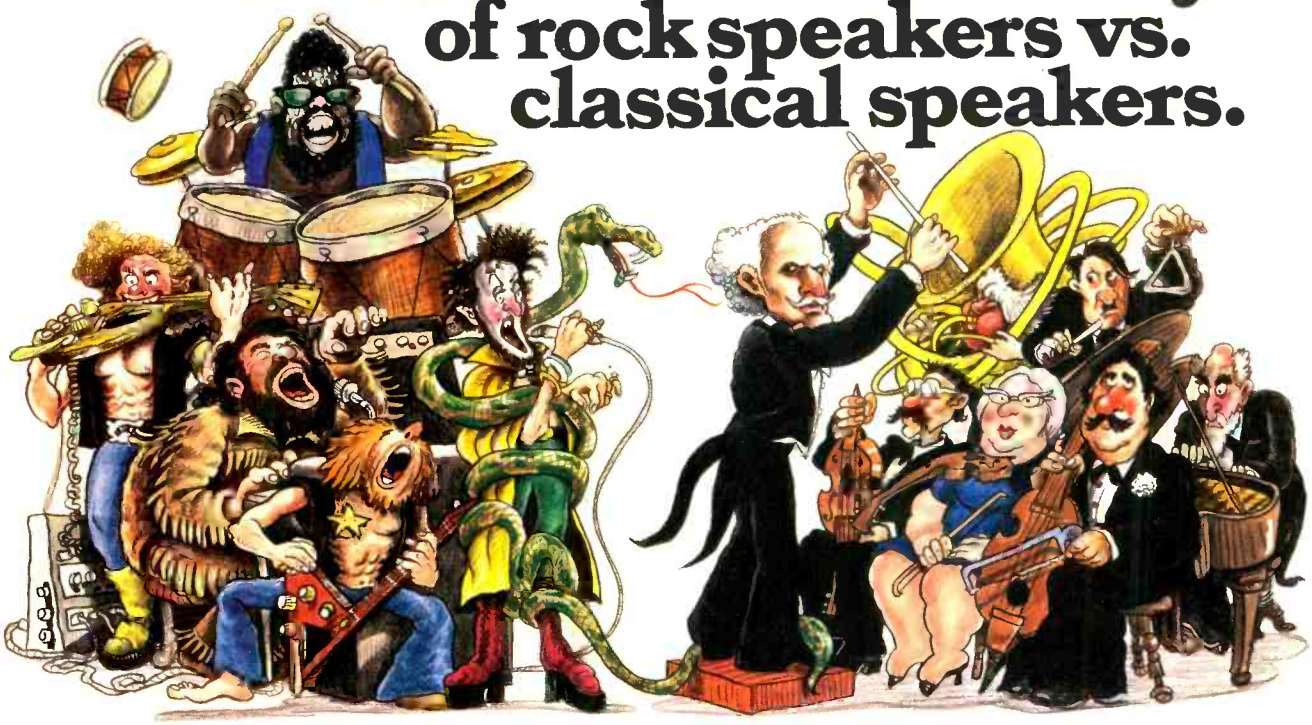
A number of interesting new items were being demonstrated in the sound rooms on the fifth floor. For example, old friend and audio pioneer Rudy Bozak was showing off a new speaker system designed for high-output Rock-music enthusiasts. In an enclosure roughly twice the size of his well-known 302 system, are mounted four 8-in. speakers of a new design, which crossover about 2500 Hz, where a curved array of light 2-in. tweeters carries on up to the high frequencies. Rudy has also entered the consumer amplifier market with a husky 150-watt-per-channel unit. The new speaker, driven by this powerful amplifier really puts out a lot of loud, but very smooth and clean sound.

In the North American Philips room they were demonstrating their amazing little motional-feedback loudspeaker, along with the Model 209 electronic turntable. Also on display were a wide variety of AKG microphones, which they represent in this country, as well as the new BK-20 AKG professional reverberation unit. I recently visited Philips in Eindhoven and AKG in Vienna, and it was a fascinating and most educational experience. I will report on this trip shortly and discuss the above mentioned products in detail.

In the Panasonic/Technics room they were demonstrating the new IC-chip demodulator. This is the IC developed by Lou Dorren of QSI and made by Signetics. Rather surprisingly, the new SH-400 demodulator is larger in size than the present discrete SE-405 unit. John Eargle, the newly installed President of the Audio Engineering Society, visited with me the weekend before the Convention and brought one of the new SH-400 demodulators with him. (John's JME Associates consulting firm has most of the CD-4 people as clients). We played around quite a bit with the unit and the quality, separation, and the sheer "discreetness" of the music on a wide variety of CD-4 recordings was truly outstanding. All that fancy new circuitry obviously makes quite a difference! A full report on this new IC demodulator will appear soon.

At the Nippon Columbia demonstration room, Prof. Duane Cooper (new President-Elect of the AES) was showing off the versatility of his UD-4 quadraphonic disc system. Mono and stereo compatibility was emphasized, and then a recording would be switched progressively through the matrix modes and finally to four-channel discrete. Very impressive,

The Rectilinear 5: end of the myth of rock speakers vs. classical speakers.



The new Rectilinear 5 is capable of playing very, very loud. Rock-festival loud. Even with a medium-powered amplifier.

At the same time, it's uncannily accurate. It sounds sweet, unstrained and just plain lifelike at all volume levels.

The temptation is great, therefore, to one-up that prestigious manufacturer who some time ago announced "The first *accurate* speaker for rock music."

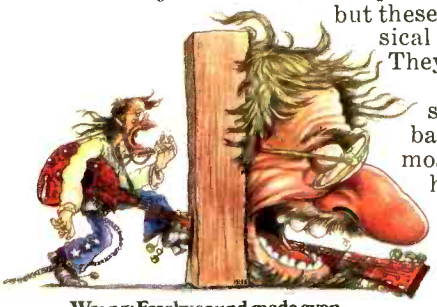
But we refuse to perpetuate that mythology. It's perfectly obvious that the Rectilinear 5 reproduces classical music just as accurately as rock. We could never see how a voice coil or a magnet would know the difference between Jimi Hendrix and Gustav Mahler.

So we'd rather use this opportunity to set things straight once and for all.

Thus:

There's no such thing as a rock speaker *or* a classical speaker. Any more than there's a late-show TV set or a football-game TV set.

There are, however, speakers that impose a hard, sizzling treble and a huge bass on *any* music. And others that round off the edges and soften up the transient details of *any* music. That's the probable origin of the myth; but these aren't rock and classical speakers, respectively. They're *inaccurate* speakers.



Wrong: Freaky sound made even freakier by the speaker.

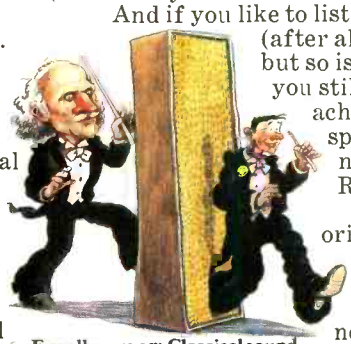
It's true that an aggressive treble and a heavy bass are characteristic of most rock music, even when heard live. It's also true that some record producers exaggerate these qualities, sometimes to a freakish degree, in their final mix of the recorded sound.

But that doesn't mean the speaker can be allowed to add its own exaggerations on top of the others.

A loudspeaker is a conduit. Its job is to convey musical or other audio information unaltered. If the producer wants to monkey around with the natural sound that originally entered the microphones, that's his creative privilege. He'll be judged by the musical end results. But if the speaker becomes creative, that's bad design.

By the same token, if some classical record producers prefer a warm, pillowy, edgeless string sound, that doesn't mean your speakers should impart those same qualities to cymbals, triangles or high trumpets. (Stravinsky's transients can be as hard as rock.)

And if you like to listen at very high volume levels (after all, that's what rock is about—but so is *Die Götterdämmerung*), you still don't need a speaker that achieves high efficiency through spurious resonances. What you need is something like the Rectilinear 5.



Equally wrong: Classical sound made vague and spineless by the speaker.

Everything in this remarkably original design was conceived to end the trade-off between efficiency and accuracy. The four drivers are made to an entirely new set of specifications. The filter network that feeds the drivers is totally unlike the traditional crossover network. Even the cabinet material is new and different.

Of course, those who feel threatened by all this fuss about accuracy and naturalness will point out that the monitor speakers preferred by engineers and producers in recording studios are usually of the zippy, super-aggressive variety.

That's perfectly true, but the reason happens to be strictly nonmusical.

"I use the XYZ speaker only as a tool," a top producer explained to us. "I wouldn't have it in my house. It really blasts at you when you crank up the volume, so that any little glitch on the tape hits you over the head. After eight hours in the studio, that's what it takes to get your attention. I know how to deal with those unpleasant highs; they're in the speaker, not on my tape."

It's easy enough to find out for yourself. Any reputable dealer will let you hear the Rectilinear 5 side by side with a "rock" or "monitor-type" speaker. Adjust each speaker by ear to the *same* high volume level, making sure the amplifiers are of good quality. Then listen. To rock *or* classical.

Then and there, the myth will crumble.

Rectilinear 5 Contemporary Laboratory Series bookshelf/floor speaker system, $299.00 Rectilinear Dispersion Base (patent pending) optional.

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and it now appears that of all things, the UD-4 system with software and demodulators, will have its first public outing on the English market. I had the pleasure of meeting Dr. Takeo Shiga, Director of the Acoustical Products division of Nippon Columbia, and Takayasu Yoshida, Deputy Gen. Mgr. of the International Trade Division. Dr. Shiga has visited the U.S. many times and has been frequent contributor to the AES Journal. A man of wide technical attainments, at this 49th Convention he was awarded the Silver Medal of the Audio Engineering Society at the presentation banquet. I had a most interesting conversation with these gentlemen, and learned that Nippon Columbia has a digital tape recorder and has actually made some commercial recordings with the unit. I have been promised some of these PCM (Pulse Code Modulation) discs and I am agog with anticipation!

If the "Unisette" and the Haeco lathe control unit were the outstanding new items at the 48th Convention, a new development from Pioneer Electronic of Japan must be reckoned as the most important item at the 49th Convention.

Four engineers from Pioneer's Acoustical Engineering Research Laboratory presented a paper entitled "Electro-Acoustic Transducers with Piezoelectric High-Polymer Films." The term "piezoelectric" has heretofore been in the audio vocabulary as applied to cheap low-fi phono cartridges utilizing ceramics such as Barium titanate and crystals such as Rochelle salts and quartz. Recently we have had the high quality piezoelectric tweeters produced by Motorola and, in fact, there was also a paper concerning them at this Convention. What the Pioneer engineers have discovered is that a high-polymer film of poly (vinylidene flouride) has piezoelectric properties, with a piezoelectric strain constant almost ten times greater that that of quartz. Evidently research into the piezoelectric potentials of various substances has been going on for quite a while. Russian workers found that wood had piezoelectric properties in the 1940's. In the 1950s highly crystalline biological substances, such as collagen, bone, and silk, showed piezoelectric qualities. In 1959, a biological high polymer, such as whale bone and tendons, were actually used in a phono cartridge!

Poly (vinylidene fluoride) is a flouro-carbon resin. The process which renders piezoelectricity to the high polymer is described in the Pioneer paper . . . "The films (8 to 30 microns depending on their applications as transducers) are uniaxially stretched up to four times the original length at 60-100°C. Then aluminum is evaporated on both sides of the film as electrodes. The films are then polarized with a high d.c. electric field at 80-100°C for about 1 hour. The process is similar to that used for piezoelectric ceramics. When a high a.c. field is applied to the film, a hysteresis loop between the applied field and the polarization has been observed." The piezoelectricity of the film is quite stable even at 100° C, and it is not affected by moisture or dust.

In essence, this Hp (high polymer) film involves the transducing functions in themselves. The application of a.c. makes the film vibrate in a transverse direction and when the film is curved, the vibration is transformed into a pulsating or "Breathing" movement. Obviously this sort of diaphragm can be used as transducers in such things as headphones, loudspeaker tweeters, and even phono cartridges and microphones. At

Wireless

It has been suggested that a perfect amplifier would be equivalent to a piece of wire with gain.
A piece of wire? First of all it would hum, so we'd have to screen it. This would increase the input capacity so we'd have to make the screening large or the conductor small. Then we would have output resistance and, if of appreciable length, we'd have inductance and termination problems as well. All in all a 303 power amplifier would be much easier.
The funny thing is; even if we had our perfect piece of wire with gain and compared it with a 303, the two would sound exactly the same no matter how carefully we listened.
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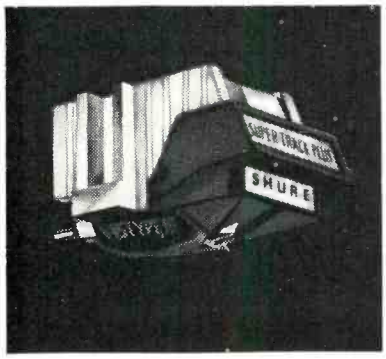
A cartridge in a pear tree.



A gift of the *Shure* V-15 *Type III* stereo phono cartridge will earn you the eternal endearment of the discriminating audiophile who receives it. What makes the V-15 such a predictable Yuletime success, of course, is its ability to extract the real sound of pipers piping, drummers drumming, rings ringing, et cetera, et cetera. In test reports that express more superlatives than a Christmas dinner, the performance of the V-15 Type III has been described as ". . . a virtually flat frequency response . . . Its sound is as neutral and uncolored as can be desired." All of which means that if you're the giver, you can make a hi-fi enthusiast deliriously happy. (If you'd like to receive it, yourself, keep your fingers crossed!)

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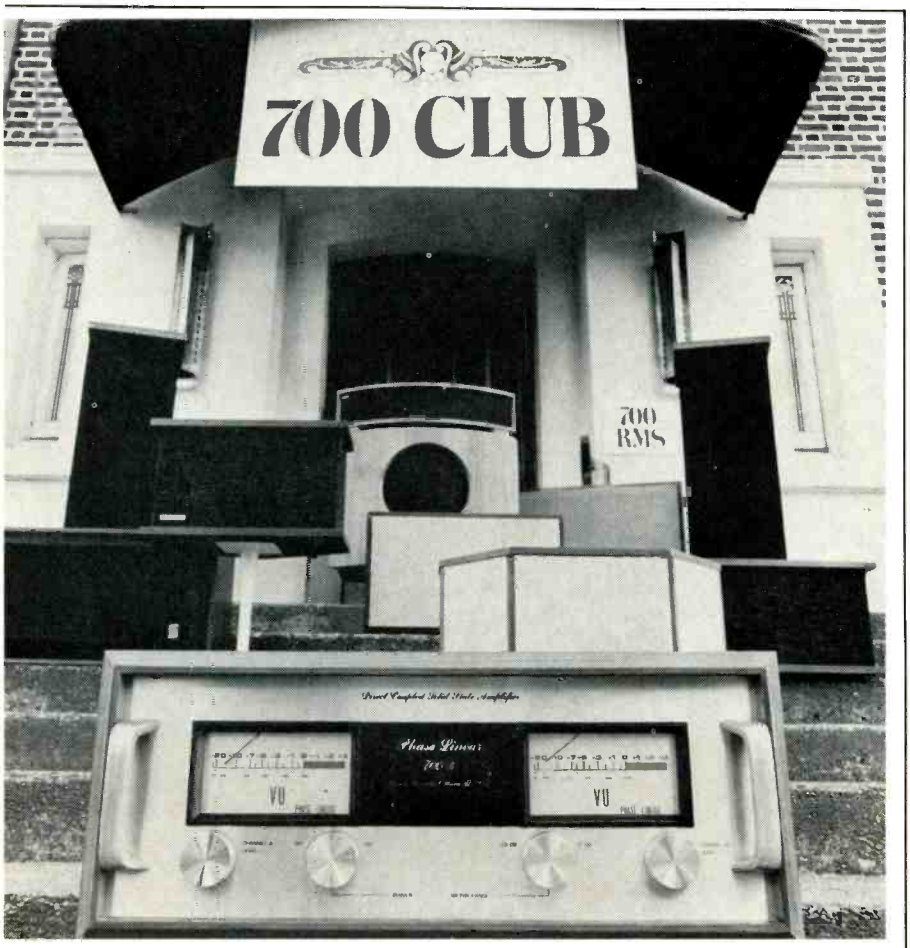
a room upstairs in the Waldorf this amazing film was being demonstrated. The first commercial application is in the form of Pioneer SE-700 High Polymer stereo headphones. These $80.00 phones weigh in at only 13 ounces and are extremely comfortable to wear. With their HP diaphragms, they can be plugged directly into 4-16 ohm headphones jacks on receivers or amplifiers. They are quite sensitive, 3 volts giving 100 dB SPL. They can take inputs up to 30 volts and 120 dB SPL without clipping. The sound was very wide range and smooth, akin to the best electrostatic phones. Also being demonstrated was a loudspeaker of the column type, with downward firing woofer. This crossed over to a circular "mid-range" HP film unit operating from 2 kHz to 7 kHz, where a smaller cylindrical tweeter carried on up to 20 kHz. The pattern of the two HP units is truly omni-directional. The sound is characterized by extreme smoothness and excellent transient response. We were also shown non-working models of a mike and a phono cartridge. I spoke to Dr. Takeo Yamamoto, Pioneer's Director of their research lab, and he stated that the HP diaphrams could just as easily be direct-radiator types as well as the omni-directional type on demonstration. When I asked him about the feasibility of making up say 4-by-4-in. panels of the HP film in multiples, a la RTR, Crown, and some other electrostatic speakers, which would permit high output with very low distortion and full frequency range without crossovers, he indicated it was just a matter of implementation. Certainly this HP material opens up whole new avenues of experiment in areas where light mass and low mechanical stiffness and the self-drive of piezoelectricity can be applied to transducing functions.

Next AES convention will be the "Golden Fiftieth," and will be held at the Cunard International Hotel in England during March of next year.

I see by the calendar that this is the Christmas issue, and I have a jim-dandy of a present for anyone who owns tape recorders and does any sort of editing. This is a new kind of tape splicer made by Nagy Research Products, Box 289, McLean, Virginia 22101. If you've ever tried to cut mylar tape with the usual single-edge razor blade, you know how tough mylar can be. It can wear out blades very quickly. Mr. John Nagy uses the same dove-tail splicing groove to hold the tape as in the familiar Editall units. But . . . his blocks are equipped with stainless-steel, self-sharpening shears. Remember to keep a bit of pressure to the left side of the lever-type shear and the Nagy will cut the mylar cleanly and as easily as a hot knife through butter *and* sharpen itself at the same time. The simplest Model 6S25, is $16.95, the more elaborate Model 25STS with a splicing tape dispenser costs $24.95. The one I have is the top of the line TS-250 and it is a real beauty. It has the tape-cutting shear, then a splicing tape dispenser mounted on a hinge, and when the desired length of tape is pulled from the dispenser, then the assembly swings down on the hinge and positions the splicing tape precisely on the two pieces of tape you want to join. Now this is hard to visualize, but there is yet *another* cutting shear that cuts the splicing tape and through the ingenious design does not cut the magnetic tape. It all works much easier than this description. With it you get professional quality splices with an absolute minimum of fuss. At $49.95 this is a super gift for the tape recorder buff.



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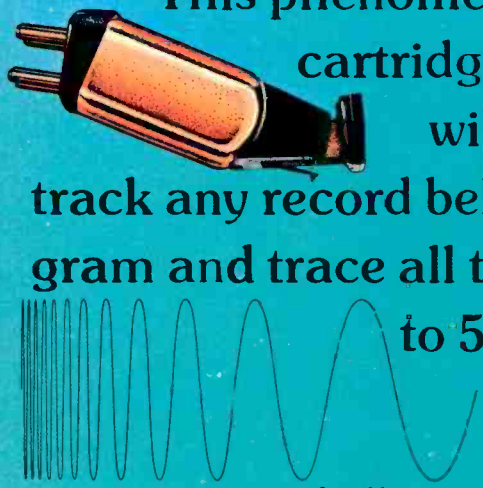

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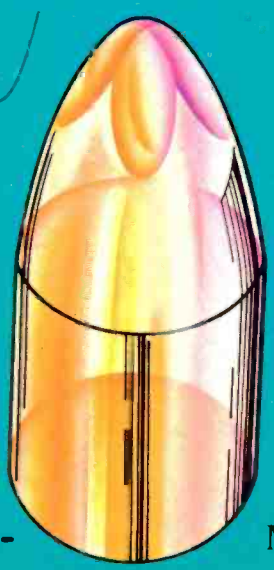

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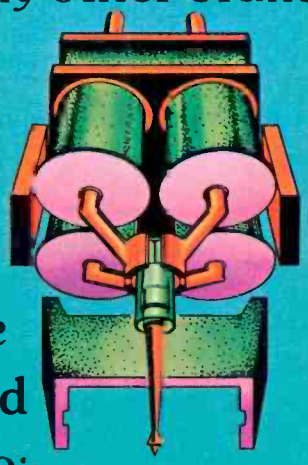

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Audio ETC

Edward Tatnall Canby

By now, my plastic, binaural friend Ispy (see last month) has become a conversation piece in my living room, except when we venture out for more recordings. I find Ispy extremely useful as the Head Between the Microphones—even though Sennheiser's tiny binaural electrets dangle just as easily from my own ear lobes as from Ispy's. The reason is that, as I quickly discovered, my head keeps moving around involuntarily. Just as nature intended. Ispy's doesn't. In binaural playback (which nature did *not* intend), the sound moves, not me. Egocentric predicament! The world revolves, all too literally, around the binaural listener.

We use *motion*, of the body and the head, to decode many details of the pairs of audible and visible signals that reach our ears and eyes. (We have a two-channel nose too, but the smell ends up mono.) You will find that it is difficult to keep your head steady, with those binaural mics attached. I keep saying that *this* time I will not move; but I forget, and in the playback there's the swing and sway again.

Imagine me, for instance, at a summer rock concert, big ski lodge with green slopes above for the audience, and me a quarter-mile up, far out in the listening crowd. Plenty loud! But at that distance, the slightest head movement is translated into a huge displacement of the sound source. So in playback the rock band zooms dizzily from side to side as though on some sort of jet-propelled cloud. Have to hear to believe. The binaural listener is *always* stationary. It is the recording which moves.

It seems that this very head motion has much to do with our normal perception of sound out in front of us, so curiously missing from the binaural playback, as described last month. CBS Labs proved this to me neatly. They have a fancy binaural dummy head with mics embedded in realistic ear canals (like the professionally available Neumann head), the mics direct-connected to phones. I put on the phones and the dummy head was moved from side to side as somebody talked directly in front of it. I was told to move *my* head in the same pattern—and immediately in my phones I heard the sound right out in front. The two motions were locked together and I was getting the message as nature intended. Unfortunately, we can't do this for binaural playback of recordings; the mics and listening ears aren't motionally tied together and so motion in either one of them is falsified.

I should add that in my own early experimenting I found that, since eyes and ears work together, the eyes also have a lot to do with locating sounds out in front. Back in 1954 I had a short movie made of a person talking (myself) and simultaneously we made a binaural recording (not even with a head—just two mics located right at the camera) of the speech. The instant we synced the two, we could hear the voice, via the binaural phones, coming straight from the screen out in front. No trouble at all. The eyes did it. So—head configurations, body motions, eye fixings, all have to do with the frontal locating of sounds in front, as of course does the plain knowledge that the sound *is* in front. Psychic power of suggestion! It helps. When these extra elements are missing, attenuated or confused, we are unable to throw the sound out where it belongs, though sidewise sounds are easily perceived via signal phase differences, plus loudness and frequency contour.

It is possible to live with a bit of unintended head motion if you are careful in your recording. I use my own head for portable recordings, now with Sennheiser's mics, in the past via my "Edward" cap with two dynamic mics fastened on each side above my ears. Works fine and frees the hands. You carry your portable recorder with one hand, use the other to fend off tree branches, briars and what-not, or to grab support if you are climbing mountains or horsing around slippery indoor swimming pools. Just don't turn your head too suddenly and things will be mostly OK. (But if you turn outdoors when a plane is passing, it suddenly reverses and flies backwards.)

A million things to record with a portable rig and no power cord. Outdoor music fests of all kinds. Games in profusion. Auctions—great! Picnics, get-togethers, walks, the sounds of city traffic, street interviews. Anything that seems utterly improbable for normal recording. Just turn the machine on and keep your head steady. You can even walk, but the playback will sound as though you were stamping up and down in one spot. No forward motion. The sounds of birds in their natural environment, off in the distance, my all-time favorite binaural subject. Auto races—terrific! A natural, since the action goes just where you want it to go, from one side to the other. In binaural, those cars really zoom past and none of the silly U-turns you get in stereo loudspeaker recordings.

Indoors, there are also portable situations. Basketball and hockey, in the stands, right from your seat. Assorted musical affairs where wall sockets are inaccessible and trailing wires unwanted. Meetings of all sorts, recorded, again, right from your place, wherever it may be. In all of this, your own head with mics in the ears is far less conspicuous than Ispy and, anyhow, he is clumsy to carry around on portable assignments.

I use Ispy for stationary recordings where a line cord is available, and I have a 200-ft. reel for extra flexibility in that respect. Just set him up where the action will be, and leave him. He'll stay put. No motion. Meetings, again, concerts, events where Ispy will not be in the way. Above all—take him to parties. My last cocktail party, with Ispy in the bushes nearby, lasted an hour and a half (3¾ ips on 1-mil Audua tape) and every minute is a joy to hear. One participant (named Joy) couldn't be pried loose from the phones for fifteen minutes. The hostess played the entire tape, so she could hear all the delectable things she missed at her own party. Note that via loudspeakers this same tape is just a jumble of voices and you would not want to listen for half a minute.

Now we get to practicalities. Is binaural recording in any way commerical? The inevitable question in our hard-nosed industry. Well, there could be binaural recorded releases. And binaural equipment, as per Sennheiser—and see below. Any hobby that fascinates the mind and the ears for cash is potentially commercial. If the right route is taken. But the wrong sort of binaural would

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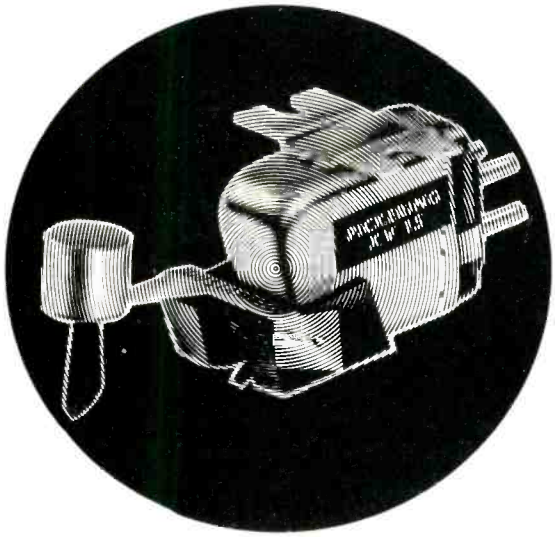

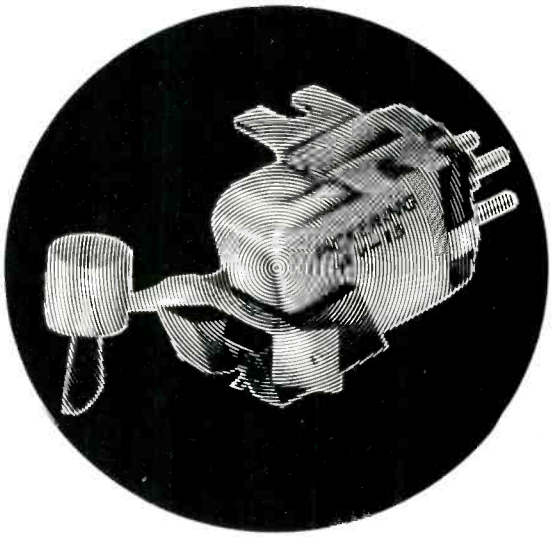

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end up a super-dud.

As for recording in binaural for commercial release, we should *never* plan to record such standard subjects as a symphony orchestra or a pop band. Sheer waste of time. True, you can get a startling binaural recording of a musical concert. But perversely, in the playback you are likely to find that the "junk" which is normally cut from the tape—talking, chairs being moved, coughs, rustlings of nearby paper (programs being turned), the noises that happen *between* the musical items, are more interesting than the music itself! It's all around you, this sort of noise, shockingly close and real, right in your ears, over your shoulder; whereas the music just sits up there and plays. If you want the music, then play a standard stereo recording.

Recordings engineers haven't worked all these years on loudspeaker-intended sound for nothing. Loudspeaker hi fi is good, an immensely subtle and highly advanced art. Binaural music recordings really have no major advantage over the sound of good modern stereo or quadraphonic, even if the phones do "surround" you—a dangerously easy catch phrase for binaural publicity. Forget it. Moreover, present "stereo" phones already give us maybe 90 percent of the impact of a specifically binaural recording of the same material, even if the standard recording was intended for loudspeakers, not phones. There is a difference, but it isn't really a *commercial* difference. Stay away.

Binaural thrives on the zany and unexpected. I put phones on my visiting brother's head, without explanation, for a bit of that cocktail party recording. You should have seen his face. At that moment some small kids had climbed into my bush to investigate Ispy and they were right at his shoulder. My brother jumped as if he were shot—Hey, there's a child right behind me! and he turned around to look at the empty space behind him. Uncanny, said he, completely baffled. What?—I didn't explain. I just smirked.

Commercial or saleable recordings, yes, *if* they really exploit the binaural quality. I have long tried (in vain) to interest the bird song recording fraternity in binaurally recorded birds. They won't even listen. They have a vested interest in those complicated parabolic-reflector mono recordings made with a truckful of expensive equipment. Nevertheless, binaural bird recordings are better, because you hear the birds—at any distance—exactly as they are heard on the spot. Again, uncanny! Somebody, some day, is going to issue binaural bird records and I trust they will ask for my help while my legs are still under me. Any birdologists interested in experimenting?

Sound effects in general, loosely speaking and on a large scale, are the most likely area for good binaural record releases (disc or tape). Anything, everything that is hopelessly difficult for loudspeaker recording. A World's Fair tour, Central Park on a Sunday, a noisy outdoor market, a boiler factory or shipyard, the Marines in basic training (assuming they'd let you). Spoken comment is easy; just talk. (Or add later via mixdown.) The binaural attention span is long, remember, and so these recordings should be in depth, at length, never those ultra-short excerpts traditional in older sound effects recordings. Hey—think of all the ethnic music and drama, the world over! 500,000 people massed together in India for a religious festival. African war games. I could go on & on. These things in binaural recordings you could really sell. Not show tunes and symphonies.

About that name Ispy. Of course I had spying in mind, as well as the famous Nixon tapes. But also all sorts of non-clandestine voice recordings such as are made by the million in mono every day. Why in Heaven has nobody yet tried to record these lively discussions in two-channel binaural? For every situation this would dramatically improve intelligibility and impact.

You don't have to be literal. You can put Ispy aside—no head needed. Just two omni mics, a few inches apart, optimally eight, though four or five will do. Miniature mics are useful but big ones are perfectly OK. Just sit them on the table, or hold them, or fasten them to some sort of modest support. Invisibly, they might go into a fancy desk stand with pen and pencil, a tiny mic at each end, or maybe a pair of book weights—a bookish "head" between. Or you could build them into a phone base, and maybe pick up the incoming signal too. Whatever you do with them, your recordings will be startling realistic. *You are there.*

I am chary of clandestine recordings, as who isn't these days. But I must observe that even the simplest binaural set-up would increase the basic communication by a huge factor. Only the telephone is beyond its use. The phone is irretrievably mono. Mr. Nixon, if I am right, used a half-track mono Uher (one-way only on the tapes) at 3¾ ips. Now suppose an exactly equivalent machine with the much more common four-track stereo configuration had been used, and there were *two* little mics hidden away at every location—just think! No "unintelligibles". No disagreement as to what was or wasn't said in the midst of the sonic interference. Every word as clear as it was in the flesh at the time. *You are there*, again. A breathtaking thought.

Enough of clandestine recording. Students! Rig up your recorders so you, too, can take down two channels of sound at lectures and the like, for binaural playback. It isn't easy (at the moment) to find a two-channel battery portable, and the Uher and Nagra IVS are for millionaires. Plug-in a.c. cassette models are everywhere, though, and the cassette is the obvious medium, even a fancy stereo cassette deck if you can find a wall outlet near enough. Mount your two omni mics from five to eight inches apart, on the recorder, on your head via a band on a hat, pinned to a sweater, fastened to a block of wood or a plastic rule—use your ingenuity. Or just hold them in your hands. Or clip them on a briefcase held in your lap. With these, you are ready to record *at any distance*, even from the back row of a lecture hall with the lecturer hundreds of feet away. If you can hear him on the spot, you'll hear him just as clearly in the recording. Phones for playback, of course. No loudspeakers. Even better for group discussions, with lots of people talking at once.

Aha—the Board Meeting! I saw a fine plush board conference room on Park Avenue in New York a few days ago. Huge table, a dozen big chairs, grand view over the city, etc. In the middle of the table was a little cassette recorder with built-in single microphone. Idiots! 'Way off-mic for every chair around the table, and total chaos if more than one person were to speak. Now, perhaps no one ever speaks out of turn in such impressive surroundings, but imagine a binaural recording, two mics, from the same spot on the table, and phones for playback at every seat. Clear, natural sound, precise and intelligble even with several people talking. Even the "off the record" jumble of conversation before and after would be perfectly understandable. *You are there*! Ho, Mr. Board Chairman, how would that be for a 100 percent improvement?

Now the clincher. Why improvise. It's time some of our audio manufacturers got into these areas with specific binaural equipment. Easy enough. The stuff is largely already on hand. Just minor modifications.

Make up an educational package, for students. An a.c./battery two-channel cassette recorder with two built-in mics, maybe at the corners

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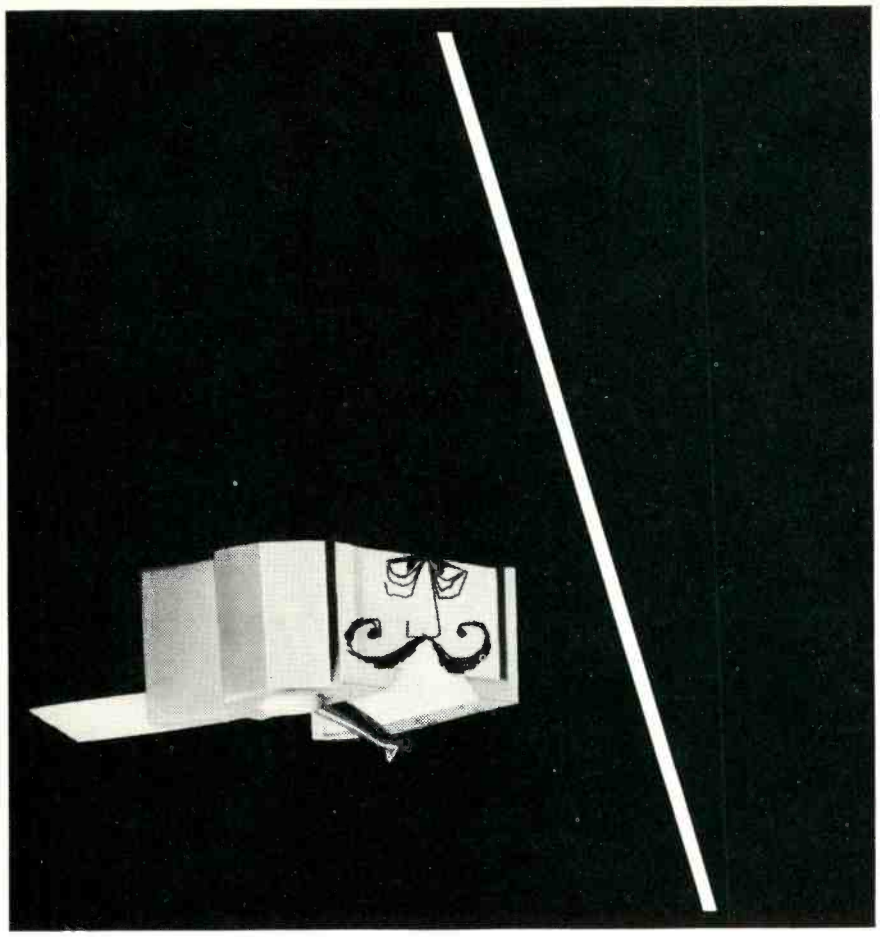

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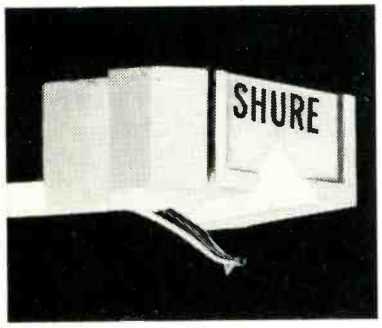

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of one end, a la Sony. Limiter circuits to control level. No speakers. Sold with or without phones. (Many students already own phones.) A fancier model could be a composite, with speaker or speakers for conventional cassette material. You could sell thousands of these to students, once the idea got around.

For the more opulent board meeting trade you could work up an impressively fancy range of Binaural Conference Recordings Systems, with trimmings galore. It *has* to be expensive, for such places. High-quality miniature mics, of course, maybe mounted on a flossy board-table centerpeice, with the firm's logo optional in gilt. A showy table recorder or, better, a side-of-the-room or remote control recorder, cassette or reel-to-reel. You name it. And don't forget a row of shiny buttons for the Big Boss to push. All this could be very impressive, but I can say with certainty that such a system would benefit thousands of management officials, corporate, governmental, educational. How about a simpler, plainer system (at a third the price) for school boards and town meetings and zoning committees and the like? It would be just as good in the sound.

A brief word about those Sennheiser Open Aire headphones, which came to me along with Sennheiser's Ispy and mics. They are excellent in sound and unique in operation. Bipolar reproducers, you understand, with slots right through to the outside—you can talk and hear straight through them. For full bass, you do not need the usual ear seal and pressure. The phones just dangle loosely, and you hear all. Light as a feather. I am enthusiastic about them for many uses, though not quite all.

They are fine for music listening *if* there is no outside noise around. (Or if you want to hear outside voices, catch the phone ringing, etc.) But you lose that "other world" privacy of the usual surround-ear phones. And in noisy situations, conversation, the inside signal is hopelessly tangled with the outside. Use them for wonderful listening when you have the place to yourself.

It strikes me forcibly that the biggest need for these Open Aire phones is professional—in the sound studio. Wow! How many years have I dangled one phone off an ear, to hear what goes on outside? Haven't you? With Open Aires you can hear in full stereo *and* communicate with your colleagues without taking the phones off. A great economizer, and Sennheiser should be able to sell a host of these to our trade, once the idea gets around. So—a merry Xmas in binaural, and a good night to all.

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Mikes of the Pre-War Era

Bob Paquette

THE HISTORY of the microphone should probably start in the year 1667 with Robert Hookes "Bull Roarer," a system similar to the two tin cans and a string that every kid interested in communication has tried at one time or another. However, while the period from 1667 to 1837 would show very little progress, the period from 1837 to 1876 would turn up experiments on practically every known microphone in use today, (some being set aside temporarily and later revived when other associated apparatus made them more feasible). The period from 1876 to 1900 spent primarily in developing loose-contact devices which evolved into carbon transmitters (or microphones) and the improving of these transmitters.

The term "microphone" was used by Wheatstone in 1827 to describe an acoustic device, by Berliner in 1877 to describe his loose-contact transmitter, and by Hughes in 1878 to describe his loose-contact devices. Aside from these few instances, the term "microphone" was generally associated with broadcasting and later with sound systems and motion pictures.

One of the first wireless transmissions of speech was in the summer of 1900. Using this as a starting point for broadcasting, we can consider the telephone transmitters of that era as our first microphones and thus establish a starting point for the development of the microphone. (See Fig. 1.). By the year 1900, over a million telephones had been manufactured and telephone parts were readily available to anyone who could afford to buy them.

The early radio experimenters used telephone transmitter elements on low-power radio transmitters but found that as they increased their radio power, the transmitter element (only able to handle 10 to 20 mA.) would get hot and burn up. The problem was the modulating of the r. f. energy with the microphone in the antenna circuit. It was obvious that if they wanted to increase their radio power using this technique they would have to develop microphones capable of handling much more current. Some followed this line and came up with some mighty unusual microphones. One early high-current microphone was made up of 25 Berliner microphones, each coupled to a metal tube and all these tubes gathering into one large mouthpiece. (See Fig. 2.) Another used a larger than usual microphone element with forced air cooling via a fan mounted under the microphone. (See Fig. 3.) A third used water cooking, and a fourth used a process of continual granule replacement. (See Fig. 4.) Some of these microphones were capable of handling currents up to 15 A. Needless to say, they were large and cumbersome. Meanwhile, others were improving modulation circuits which eventually made it possible to use telephone transmitter elements in the same way that they were used in telephone circuits. (See Fig. 5.)

Western Electric was the largest manufacturer of telephone apparatus and consequently the most qualified to develop and improve microphones and associated apparatus. From 1915 to 1920 the most popular microphone was probably the W. E. candlestick telephone, less receiver. It wasn't too long before they removed the receiver hanger assembly and replaced it with a push lever, giving us a push-to-talk desk microphone. (See Fig. 6.) This same phone was further modified by removing the base, shortening the stem and installing a plug in the end of the stem with a hole for the cord, thus giving us a hand mike. (See Fig. 7.)

One of the first microphones designed specifically as a mike was the Western Electric D14298 or MW10B. (See Fig. 8.) The patent for this mike was applied for in 1913. It was designed to Army Signal Corps specifications for air-to-ground communications. Since it was to be used in open aircraft, it was designed to be directional, or as the advertising of that day put it, "will not detect any sound waves except those directed straight into the microphone." This microphone was used during WW I and later became a surplus item. Many radio amateurs, experimenters, and early broadcasters modified them by removing the breastplate and straps and using them as hand mike or making a small base and using them as desk mike.

Magnavox produced a similar mike for Navy aircraft around the same time and later advertised it as "a notable win-the-war invention." Their advertising quoted Acting Secretary Roosevelt as saying "a special feature of the telephone sets is the anti-noise microphone, which is so constructed that the engine noises are not heard. This is accomplished by having the back of the microphone open. The exterior sound waves strike the back as hard as the face of the diaphragm, and therefore the effect is neutral. The voice waves strike only the face of the diaphragm, and the radio sets receive enough effect to modulate the transmitted wave."

The invention of the Audion and the development of the audio amplifier between 1906 and 1915 resulted in more experimentation on previous ideas and new microphones began to emerge. In 1916 the condenser and in 1917 the crystal microphones were developed. The crystal at this time was a very specialized microphone for an underwater project similar to sonar. The first commercial use of a condenser microphone was in the sound system setup for President Harding's Inauguration on March 19, 1921. This was the Western Electric 394 condenser element, later to be used in their 47A, 7A, 8A, 9A, and 10A amplifiers and housings. (See Fig. 9.)

The carbon microphones used up to 1920 were single-button microphones. In 1920 Western Electric introduced a double-button carbon mike with a stretched diaphragm,

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***Editor's Note:** Mr. Paquette is assembling a history and museum of microphones, and presently has more than 300 different models in his collection of U.S.-made mikes. He also collects literature, catalogs, spec. sheets, etc. relating to mikes, and he believes that his collection is the most complete in the country. He welcomes offers of any additional microphones or literature, and he can be contacted at Select Sound Service, 443 No. 31st Street, Milwaukee, Wisc. 53208.

which improved sound quality, as frequency response was extended, resonant peaks smoothed out, and harmonic distortion reduced considerably. One minor disadvantage was the reduced sensitivity which required a little more amplification. The first model was the 373W, followed by the improved 387W and 387. Then came the 389 which was smaller in size and fit into a candlestick stand. This was called the 1441CJ announce mike. The last mike in this group was the 600A, identical in size to the 373 or 387, and with more improvements. (See Fig. 10.) A Model 41A hiss filter fastened on the back of any of these three. Aside from the 389, all were spring mounted in rings, though housings were available to give them a more finished appearance. The most popular of these were the 1A and 1B housings. The only difference between these two is that the 1A came with a three-wire cord attached. (See Fig. 11.)

The early carbon mikes had many problems—if they were moved or jostled while in operation, arcing would

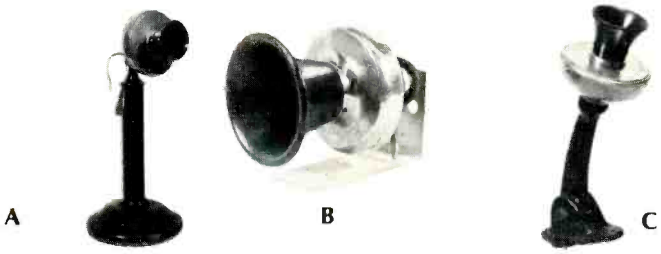
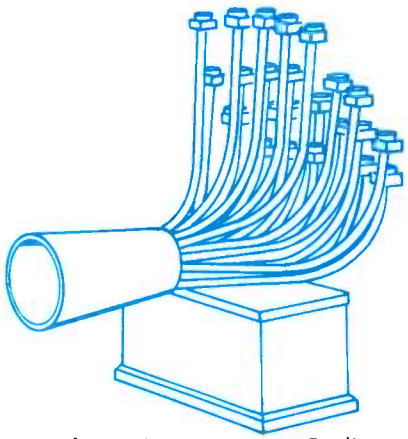


Fig. 1—A, American Bell Telephone Model 299 on 1915 candlestick stand; **B,** Berliner transmitter, and **C,** Century Telephone Construction Co. mike, off of box telephone.

Fig. 2—C. Lorenz Co. multiple transmitter (after Wireless Age). *Below left.*



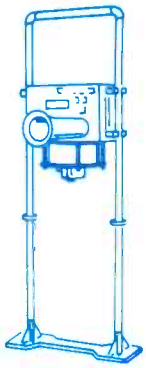


Fig. 3—Heavy-current Berliner microphone transmitter with fan cooling (after Wireless Age). *Above right.*

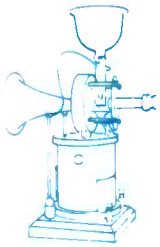


Fig. 4—Scheidt-Boon Marzi high-current microphone transmitter (after Wireless Age).

Fig. 5—Three mikes capable of 2½ amp currents: **A, B,** Electra Voice, and **C,** Universal High-Power Telephone Co.

Fig. 6—Western Electric 323 on candlestick stand with push-to-talk switch.

Fig. 7—Western Electric 323 hand mike.

Fig. 8—Western Electric D14298 or MW10B.

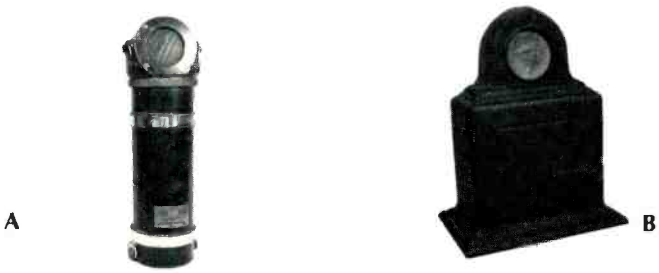


Fig. 9—A, Western Electric 47A condenser, and **B,** Western Electric 7A condenser.

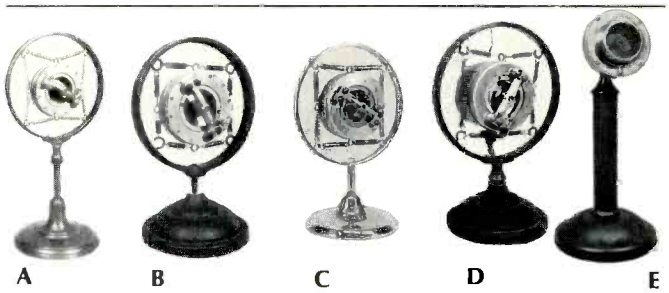


Fig. 10—Five Western Electric spring-mounts: **A,** 373W; **B,** 387W; **C,** 387; **D,** 600A with hiss filter, and **E,** 1441CD with 389 element.

Fig. 11—Western Electric 1B housing.

occur within the granules and against the diaphragm, the result being a deterioration in performance and a shorter time between granule and diaphragm replacement, and in extreme cases the carbon granules could fuse together rendering the mike inoperative. If the mike sat in one place too long or if the granules absorbed moisture, they would pack and result in very little output. The studio engineer would usually tell the people using the mikes to keep an eye on him in the control room, so that if the mike stopped working he could signal them to give it a few quick shakes to loosen it up.

These mikes could only be used in a vertical position and since the diaphragm was easy to get at in most units, they had to be handled with care. The diaphragm, being stretched and under tension, could split if touched with a sharp object. The diaphragms in the Western Electric double-button mikes were made of two-mil duralumin and the condensers used one-mil duralumin. To replace the diaphragm and carbon granules in a mike of this type took a minimum of 2½ hours. The double-button carbon mike was the standard of the broadcast industry from 1921 to about 1931 and many were still being used up to 1940.

In the early 20s Western Electric was the only company producing quality microphones and purchasing one from them was quite difficult. They sold complete systems which included a mike or two, but rarely sold an individual mike except under special circumstances. The situation seemed to get worse in the mid-twenties when Western Electric developed their talking motion picture and a new method of electrical recording and reproduction of phonograph records. They licensed Warner Bros. in the motion picture process and Columbia and Victor in the recording process. This gave them the opportunity to lease practically everything they produced.

Dr. Phillip Thomas of the Westinghouse Laboratories developed a glow discharge mike in 1923 and used it on station KDKA for a while. Apparently there were problems with it since it never got into production.

In about 1923 or 1924, RCA was selling a double-button carbon mike made by Kellog, which wasn't as good as the Western Electric's but was probably the next best available. Kellog also sold them to others and helped to fill the demand for mikes.

In 1924 RCA introduced their 3A box condenser mike, which did not measure up to the quality Western Electric condensers but was available to anyone and it became very popular. It's main problem was that it was very noisy. The original design used 199-type tubes that were inherently microphonic, creating a very noisy mike whenever it was touched, bumped, or vibrated. This problem was solved when they redesigned the circuit to use UX864-type tubes. The new model 4A was a vast improvement over the 3A but still didn't meet the standards of the Western Electric condensers with their quieter liquid resistors and more stable gas-filled condenser head. (See Fig. 12.) Replacing diaphragms in these early condensers was also quite a project, particularly in the Western Electric which took about six hours.

Between 1920 and 1927 Western Electric and RCA were the only big companies producing many microphones, but these were nowhere near enough to supply the expanding broadcasting, sound and recording industries, not to mention amateur radio people. The prices paid for these mikes were higher than the amateur or sound shop could afford, so microphone companies popped up all over to supply the demand for lower-priced mikes. Ellis, Shure, Electro-Voice, American, Universal, Gavit, Astatic, Jenkins Adair, Remler, Amperite, Lifetime, Bruno Gibbs, Seco, Amplion, Brush, Bud, Connecticut, Chrisell Acoustic, Electro Acoustic, Federal, Frost, Miles Reproducers, Radio Receptor, Sampson, Telephonics, Thomaston Labs, Transducer Corp, Turner, and E.F. Johnson were some of the companies that produced mikes in the later 20s and the early 30s. Most of these companies produced a cheaper version of the same types of microphones already being marketed.

The year 1931 was the start of a new era in microphone development, as Western Electric introduced their moving-coil microphone, RCA their ribbon velocity microphone, and Brush their Rochelle-salt crystal microphone.

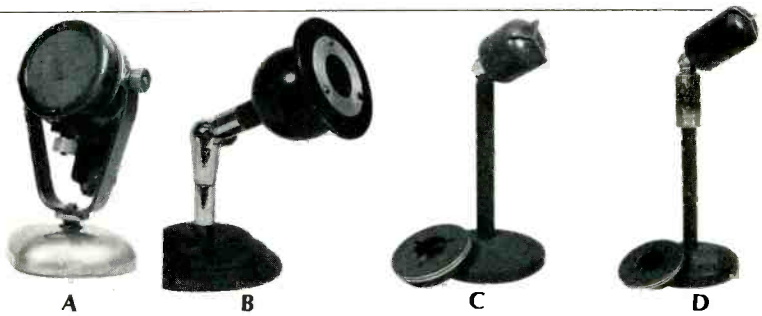
Western Electric's first moving coil mike was the 618A, followed by their 630A, 632A, 632C, and 633. Because of their appearance, the 632C and 633 were nicknamed the "saltshakers." (See Fig. 13.)

RCA's first ribbon velocity was their Model 44, a large cylinder containing a preamplifier with the ribbon element hanging from the bottom of the cylinder in a separate, perforated housing. This unit was first sold to their photophone franchise holders for motion picture work. One trade journal of that time said, "the new mike will give to photophone franchise holders at West Coast studios the same advantages for noise elimination as those accorded Western Electric when the latter, in December, completed development of its new 'noiseless recording system', now in use at practically all Hollywood studios using Erpi sound systems." This was in reference to the Western Electric 618A moving-coil mike.

The 44 was followed by the 44A and 44B and later by the 44BX. These three models were excellent microphones and were probably the most popular mikes used in radio in the later 30s and throughout the 40s. The 44A was antique bronze, the 44B was black with a chrome screen and chrome

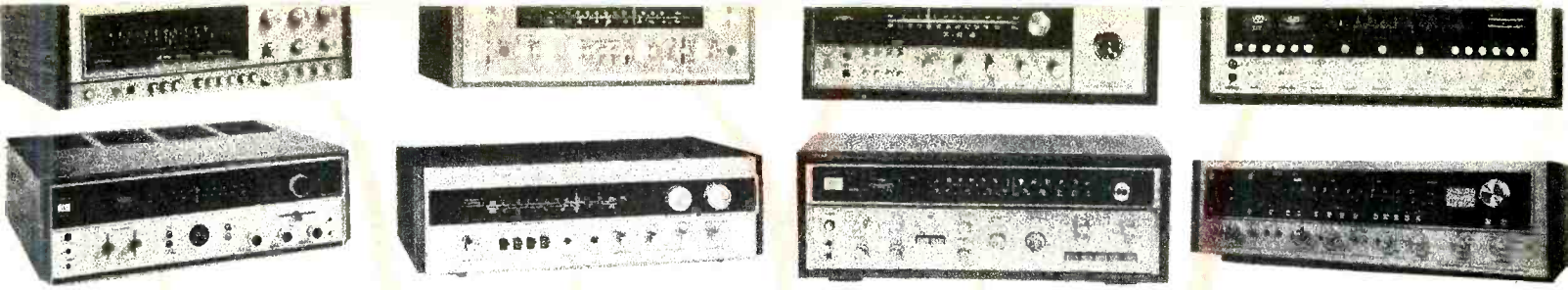
(Continued on page 28)

Fig. 12—RCA 4A box condenser on announce stand.



A B C D

Fig. 13—Four Western Electric mikes: **A,** 618A; **B,** 630A; **C,** 632C, and **D,** 633A.



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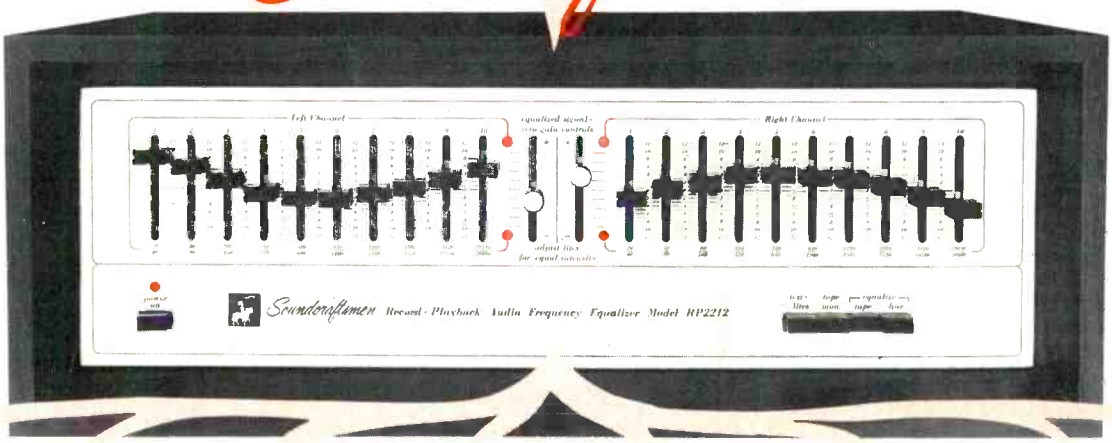

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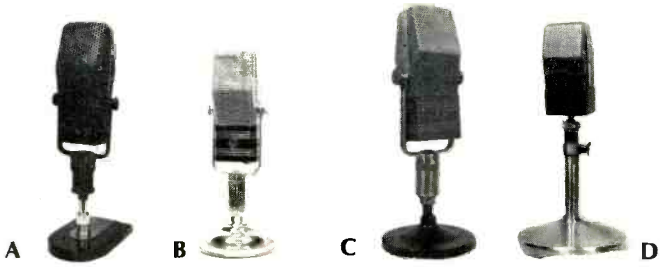

Fig. 14—Four RCA mikes: **A,** 44A; **B,** 44B; **C,** 44BX, and **D,** 74B.

fittings, and the 44BX was biege with chrome screen and fittings. All were quite large and weighed 8½ lbs. each. A later model, the 74B, was much smaller and only weighed 2½ lbs. This mike was referred to as the "Junior Velocity" and its response characteristics were much less than the 44 series. (See Fig. 14.)

Our very definite advantage of the ribbon mike was its figure eight or bi-directional pickup pattern. Being dead from the sides, top, and bottom, this mike could be positioned to eliminate studio sounds, noises, and reverberation. In motion picture work camera noise was always a problem, and special "blimps," sound proof booths, and baffles used to quiet these sounds could be eliminated by positioning a ribbon mike properly.

In 1934 RCA introduced the 77A microphone which went a step further in directional characteristics and is the first uni-directional or cardiod pattern microphone. By using a ribbon clampled down in the center and using the upper half of the ribbon with the back enclosed and controlled, this upper ribbon became a pressure type mike. The combination of the pressure and velocity made the cardiod pattern possible. (See Fig. 15.) This mike was followed by the 77B, 77D, and the 77DX, each of these mikes being an improvement over the previous model. (See Fig. 16.) Another old ribbon velocity mike made by RCA was the 1-0001. Specifically designed for boom use, it was a sphere about 7-in. diameter and nicknamed the "skunk mike" as it was all black with a white stripe down the perforations in the case.

Although the ribbon mike had very definite advantages over the moving-coil type, they also had some disadvantages. If you tried to work the mike too close, it would become very boomy due to proximity effect. In later ribbon mikes, equalizers were built in with a switch labeled "voice" —"music" to take care of this problem. It also required a special windscreen if it was to be used on a boom or moved while being used. For the same reason, outdoor use of this mike without a windscreen was virtually impossible if there was any wind at all. A light breeze across the ribbon would create thunder in the speakers rivaled only by a good storm. For public address work where you had little control over the user, it was a disaster. Asking people not to blow into a microphone seemed to be more of a challenge than a request. Invariably, some one would blow into it and destroy the ribbon.

Replacing a ribbon is not a hard job but it is an exacting and tedious job, requiring a clean area with no air movement (including heavy breathing), non-magnetic screwdrivers and tweezers, a little spit, a toothpick, a magnifying glass, scotch tape, and plenty of patience.

RCA introduced another type microphone in 1935; it was called an "inductor" mike and the first model was the 50A. This was actually a pressure mike but used a type of construction very similar to the ribbon. A single 0.010-in. conductor was rigidly coupled to a diaphragm and located between the poles of a permanent magnet with its length perpendicular to the magnetic lines of force. Its performance was quite similar to the moving-coil mikes. It was later followed by the SK45, also an inductor mike used primarily for announcing or paging. (See Fig. 17.)

Crystal mikes were of two basic types, the diaphragm and the sound cell. The diaphragm type was economical to make, had high output voltage and fair frequency response. The sound cell was more expensive, had low output voltage, and an excellent frequency response. The diaphragm type became very popular with amateurs and P.A. operators. The sound cell types were used more in radio stations and recording studios; however, having extremely high impedance, they were soon considered obsolete for studio use. The losses in high impedance cable were critical as cable length increased and became more susceptible to hum and noise pickup. The studios began to standardize on low impedance microphone input circuitry, which meant that a crystal mike needed a matching transformer. As this occurred, the quality of the moving coil or dynamic mikes improved and the crystal mikes began to disappear from the studio scene. Another factor influencing this disappearance was that the crystal mikes were more fragile and susceptible to heat and moisture. Many microphones were destroyed on automobile seats with the sun shining into the auto, or in the sun on a windowsill in the studio. Also, if a crystal mike was stored for a long time in a very humid place, the crystal would begin to decompose.

The Brush Development Co. held all the patents on crystal devices and produced a line of microphones. (See Fig. 18.) They also licensed many other companies to use their patents so crystal mikes were very plentiful for many years.

RCA's uni-directional mike solved many noise and feedback problems, and the other mike companies had to follow

(Continued on page 92)

Fig. 15—RCA 77A.

Fig. 16—Three RCA mikes: **A,** 77B; **B,** 77D; and **C,** 77DX.

Fig. 17—Two RCA mikes: **A,** 50A, and **B,** SK45.

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Directory Addenda

IN THE receiver section of our October product Directory, the Sherwood and Kenwood lines were unfortunately placed together. We apologize to our readers, as well as to these firms, for the inconvenience, and we herewith present the two receiver lines as they should have been shown.

A number of prices were incorrectly listed in the Sony Corp. lines of receivers, turntables, and speakers. The correct prices for the receivers are: Model SQR-8750, $669.50; Model SQR-4750, $479.50, and Model STR-6036A, $229.50. Correct prices for turntables are: Model PS-2251, $429.50, and Model PS-1100, $99.50. Correct prices for the speakers are: Model SSU-1800, $199.50 each; Model SSU-1600, $149.50 each; Model SSU-1400, $109.50 each; Model SSU-1200, $79.50 each, and Model SSU-1100, $99.50 per pair.

Toshiba's Model ST-500 tuner, which was listed with an incorrect price, actually retails for $239.95.

The forms covering Fisher Radio and Great American Sound products were received too late for inclusion in the Directory, and we herewith present those firms' data. Address for them are:

Fisher Radio Corp.
11-40 45th Road
Long Island City, N.Y. 11101

Great American Sound Co.
8780 Shoreham Dr.
West Hollywood, Calif. 90069

(Continued on page 65)

Receivers

	MODEL	RMS Power/chan., W	THD at rated power, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz–kHz	1-watt freq. resp., Hz ± 2 dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	IHF sensitivity, µV, Stereo	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %, 1kHz	THD stereo, 100% mod., % 1kHz	Stereo sep., 1000Hz, dB	Tuning indicator	Alt. chan. selectivity, dB	No. of meters	AM band?	Dimensions, w x d x h, in.	Weight, lbs.	Price	Notes
SHERWOOD	S-7244	20*	.6	.6	.25	14-50k	20-20k ±1	70	3.0	100	1.9	1.5	20-15	.25	.5		MTR	55	1	Yes	18¾ x 15¼ x 5¼	40	499.95	*20-20k 4 channel SQ Logic
	S-8900A	60*	.3	.3	.06	7-60k	20-20k ±1	70	1.5	100	1.7	1.9	15-20	.15	.3	40	MTR	65	1	No	17 x 14 x 6¼	30	449.95	*20-20k
	S-7900A	60*	.3	.3	.06	7-60k	20-20k ±1	70	1.5	100	1.7	1.9	15-20	.15	.3	40	MTR	65	1	Yes	17 x 14 x 6½	30	479.95	*20.20k
	S-7310	38*	.5	.5	.08	7-50k	20-20k ±1	70	2.0	90	1.8	1.2	20-15	.25	.5	40	MTR	65	1	Yes	17½ x 13¼ x 5¼	35.5	369.95	*20-20k
	S-7210	26*	.8	.8	.1	8-50k	20-20k ±1	70	2.0	110	1.9	1.4	20-15	.3	.6	40	MTR	65	1	Yes	17½ x 13¼ x 5¼	28.5	299.95	*20-20k
	S-7110	17*	.9	.9	.1	9-50k	20-20k ±1	70	2.0	90	2.0	1.5	20-15	.4	.6	40	MTR	60	1	Yes	17½ x 13¼ x 5¼	27.5	229.95	*40-20k
	S-7010	8*	1.0	1.0	.20	15-50k	20-20k ±1	70	2.0	80	2.8	4.0	20-15	.6	.8	35	MTR	50	1	Yes	17½ x 13¼ x 5¼	27	179.95	*40-20k
KENWOOD	KR-7400	65	0.3	0.3	0.07	10-35k	10-40k ±1	70	2.5	250	1.7	1.3	20-15k	0.3	0.5	40	Mtr	80	2	Yes	19 x 13⅜ x 6	31	519.95	Dir.-cpld. amp.
	KR-6400	52	0.3	0.3	0.07	10-30k	10-40k	70	2.5	250	1.9	1.5	20-15k	0.3	0.5	35	Mtr.	65	2	Yes	19 x 13⅜ x 6	30	449.95	Same as above.
	KR-5400	37	0.5	0.5	0.07	10-30k	10-40k	70	2.5	250	1.9	1.5	20-15k	0.3	0.5	35	Mtr.	65	1	Yes	19 x 13⅜ x 6	25.4	379.95	Same as above.
	KR-4400	27	0.5	0.5	0.08	10-30k	10-40k	70	2.5	160	2.1	2	20-15k	0.4	0.6	35	Mtr	55	1	Yes	19 x 13⅜ x 6	22.3	299.95	Same as above.
	KR-3400	22	0.8	0.8	0.1	10-30k	10-40k	70	2.5	160	2.5	3	20-15k	0.4	0.6	33	Mtr.	45	1	Yes			259.95	Same as above.
	KR-2400	16	1	1	0.1	10-30k	10-40k	70	2.5	160	2.5	3	20-15k	0.4	0.6	33	Mtr.	45	1	Yes			219.95	Same as above.
	KR-1400	10	1	1	0.5	25-30k	25-35k	70	2.5	160	2	3	20-15k	0.4	0.6	33	Mtr.	45	1	Yes			179.95	
FISHER	634 4-chan.	34	.5	.8		20-20k		65	2		1.8	1.2	10-15k	.3	.4	38	2 Mtrs.	70	2	Yes	23 x 16 x 7		799.95	Built-in CD-4 & full-logic SQ.
	434 4-chan.	15	.8	1		20-20k		65	2		1.8	1.2	10-15k	.3	.4	38	Mtr.	60	2	Yes	20 x 13 x 7		599.95	Built-in logic SQ & CD-4.
	334 4-chan.	10	1	1		20-20k		65	2		2.0	1.5	10-15k	.4	.5	35	Mtr.	55	1	Yes	20 x 13 x 7		499.95	Built-in CD-4 & SQ.
	234 4-chan.	10	1	1		20-20k		65	2		2.0	1.5	10-15k	.4	.5	35	Mtr.	55	1	Yes	20 x 12 x 7		3.49.95	Built-in SQ.
	222	16	1	1		50-20k		67	2.5		2.0	2.0	10-15k	.5	1	35	2 Mtrs.	50	2	Yes	18¼ x 13 x 5⅜		249.95	
	122	11	1	1		60-20k		63	2.5		3.4	2.6	10-15k	.7	1.5	35	Mtr.	35	1	Yes	16¾ x 11 x 5¾		199.95	

(Continued on page 65)

Series 70 Recorder/Reproducers

When you've got more talent than money

TASCAM Series 70 recorder/reproducers were designed for people who've outgrown high-end consumer audio products but can't afford full professional studio gear.

Whether you need single, two or four channels, you define the Series 70...it doesn't define you. Your choices are expanded instead of restricted without paying a performance penalty.

The versatile Series 70 electronics come in two versions, one for direct recording and one for use with a mixing console like our Model 10. Either way you'll find uncommon quality and reliability.

Series 70 recorder/reproducers. When you've got more talent than money.

TASCAM CORPORATION

5440 McConnell Avenue
Los Angeles, Calif. 90066

Check No. 42 on Reader Service Card

Mike Technique And Sound Effects

A. R. Jourdan

When AUDIO's editor asked this amateur sound collector to do an article on microphone technique and sound effects, the first thing to do in preparation was to look up the word "technique" in the dictionary. After all, it does help to know what it means—which is "the best way of getting results." Actually the article is the product of some long years as a hi-fi bug and of the knowledge shared by some nice people enroute.

Good recordings were made during the early 1900s on both cylinders and discs—even before the use of electrical recording commercially. The carbon mike has a long history, as have the dynamic and condenser mikes, and naturally they have undergone some changes over the years. The first electrical recordings were made with the telephone units. Kevin Brownlow, in this book *Parade Gone By* (Knopf, $14.00), mentions a cutter invented in 1909 by Arthur Kingston for French Pathe which allowed direct talkies to be made. It used a series of microphone relays for driving the aerophone (compressed air) unit in the speakers. Cameron, in his books on movie projection, has given considerable data on the wide range of ideas for both films and sound.

Although the triode was invented in 1906 and Bell began development of it for amplifiers after 1912, speech had been transmitted over the Atlantic to Scotland and between Newfoundland and Ireland. The mike used in these transmissions had platinum electrodes, was cooled by a water jacket, and had a load current of 15 amperes. These were installed almost in the antenna line. There's more about all this in Read and Welch's *From Tin Foil to Stereo* (Sams /Bobbs-Merrill).

Acoustically, the first recording studios were considerably better than the radio stations, and conversion to electric recording was done quite easily. It was necessary to improve on the spring-wound and gravity-feed turntables, although the latter could be taken on field work. Some wax lathes were taken in large trucks to do effects as well as on-location recording of orchestras. It is probable that more of this was done in Europe than in the United States if you exclude the Library of Congress and some musicologists.

The early radio stations were quite primitive, but in due time grew to a size proportional to the present-day TV stations. It was not unusual for the average station to have more than one studio—at least one capable of holding a modest-sized orchestra and actors. At the present time, the station may be little more than one an-

nouncing booth with a few turntables and tape-playback facilities. Any spare space is most likely to become offices, since remotes are taped. If the station doesn't have a network feed, it is more likely to use the remotes including local churches, which are sometimes of dubious quality, due to conditions beyond control, and occasionally a thankless job.

Vital to a small station's good management is considerable know-how in many fields. For example, until recently there were only two libraries of sound that could be fed directly through the console, if you exclude transcriptions and tapes, those from Standard Radio and Silver Masque (Engineer). The usual method is to have two turntables and three pickups for the ten-in., 78-rpm discs. Larger studios would have more turntables, placed in units or groups, and possibly some workable cueing procedure. Tape is used presently, and it is much easier to work with unless cross-mixing is needed.

Prior to the use of recorded effects, studios relied on gimmicks and for voice such men as Brad Barker and Don Bain. More recently, of course, TV has given birth to such "voice personalities" as Mel Blanc and Joanne Worley, and "voice overs" are a standard practice in TV advertising to combine the "proper" voice and face. Such skilled mimics have been around for quite some time, and a Boston paper of the 30s described a Count Cutelliani (sp?—Ed.) who used some ten gimmicks for effects.

The first professional library of 78s was made by the late Harry Gennett at the suggestion of Thomas J. Valentino. Gennett came from a Richmond, Ind., family which owned Starr Piano, and Valentino was a piano tuner at the time. After setting up the East Coast offices in 1932, Valentino handled much of the work there. Gennett also developed a new series called Speedy-Q, which were much better than the first set and had fewer rejects on auditions. These carried a change of sound, or cue, on the spiral connecting a sound effect. A panel truck was used to go on location, and a sound film process, which could reproduce up to 8 kHz, was used for the master track. The series was stopped at the beginning of WW II, and Gennett's passing in the late 50s ended hopes for a revival. In fact, most of the Richmond discs, along with those from RCA and Columbia, stopped at that period, though a few did continue. Valentino started his own Major series in 1936 and began importing EMI mood music prior to the war. Postwar, he commissioned his own mood music for mastering in New York City studios. He

also began conversion of his 90 effects discs and mood music to LPs in the 60s. This mood music, taped in Europe, possibly was under the composer's direction. The discs were mastered by Rein Narma at Gotham studio with a new amplifier cutter and had impressive sound. At present Valentino is retired, though his three sons continue the business.

Charles Michaelson took over the Gennett series and the Standard Radio library after Alex Sherwood retired. He was written up in the Saturday Evening Post during the late 50s and sold the EMI series for a while before giving up effects to go into TV films. I had read of Valentino and Masque Engineering in a book on radio production, and he and I had quite a few interesting chats until 1967 when I stopped going to New York City. Masque now operates under new ownership and supplies effects and sound gear for stage plays; Standard Radio works from the West Coast only. Valentino does have some special pressings for some plays, including music for The Glass Menagerie.

The RCA set, mentioned above, includes some 44 records, with the same sounds on each side of the disc, and sold for $1.50. Sound quality of the material was only fair to poor. The Columbia series, all made in England, had 28 records and was a little better in sound quality.

Sad to say, the pioneer Gennett series of discs were not up to the standards of the Speedy-Q though they were produced by the same man. Speedy-Q derives its name from the fact that the sound on the disc is not interrupted by the spiral connecting the changing sections but that the visual cue point, the spiral, also has sound. The first truly hi-fi discs were brought out by Masque Sound, and their trains and boats are very good. Tom Valentino used the Fairchild 199 lathe and RCA sound on film in a station wagon to go on location. His recording of news effects was spotted by the Dick Tracy cartoonist, Chester Gould, and used for a story line.

Valentino once told me that he had spent some $10,000 doing a chime series together with organ music at the Collegiate Church with Dr. Norman Vincent Peale and that it cost him $1,000 to find the right place for the microphone. I later went to visit the late Col. Richard Ranger at his Rangertone plant in Newark, N.J., on the day I received some fresh vinyl pressings from Valentino. While in the plant I saw the process of making those chime records (from the Rangertone organ) on a Presto cutting table. The disc was flowed on soft and had a blue color and was baked to a golden yellow after cutting.

There are a number of effects discs, besides Valentino's 16, including 15 from Audio Fidelity (there is no number 13), all quite well done; 14 from Elektra plus a three-disc set titled "Most Asked For Effects" and another three-disc set of mood music; three discs from Cook; nine from Offbeat, and some on Folkways including two of mood music and one of novel space sounds. Quite a few discs on other labels could be called effects recordings. These include Philips "African-Asian Music," the Monitor disc on China (from a documentary film), one from Columbia on Scandanavia, and some from Capitol and Decca on Europe, Mexico, and Japan. Not too long ago I counted 28 bird discs, 24 of Indians, and some 33 train recordings listed in the Schwann catalog; there are probably some others around.

Mike Placement & Use

Regardless of location, we have two rules to start with in regard to placement of mikes. The mike is placed off center of the studio and toward the dead or dull side. The usual height is five feet and working distance—to start off—is about two feet. The mike least favored under these conditions would be the omni. The use of a cardioid or figure eight mike allows placements to be made to maximum advantage, supplemented by sound baffles, rotating vanes for acoustical control, or even isolation booths, as for a singer with orchestra. The present-day mikes vary from small hand-held or even smaller lapel types to shotgun units for remote pick-ups and mikes on long booms to follow the action. Especially useful are the current condenser types which allow more mikes per studio thanks to their cardioid "spotting" of individual instruments.

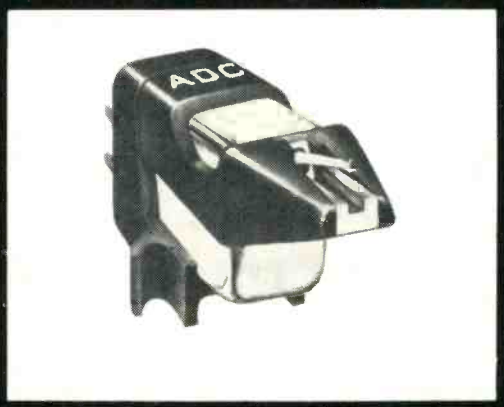
The usual range of distance and height for a 25-man orchestra is five to seven feet, on up to 15 to 25 for a full symphony. If a single unit is used, the mike is often near the conductor or at least in front and not too far away. Placements for solo or individual instruments seem to relate to the sort of musical instrument and its sound level, and the mike generally faces the source of the sound. In strings, this is the F hole, the bell for woodwinds or brass, and the soundbox of a harp. Woodwinds and treble strings can be miked at two feet, some further away, while brass should be further still, about six to 10 feet. A compromise for a grand piano is eight feet with the top open. For a church organ, the distance should be half the length of the church if it is large, but much depends on the musical scoring and the choice of stops used. If there are too many stops in use, the sound will be muddy. If the organ is played at too high a volume, the echoes will be shrill.

AUDIO has carried several informative articles on mikes in recent years, including Jim Long's in Dec., 1972, to Feb., 1973, and D.L. Josephson's in Dec., 1973, and July and Aug., 1974. There is one book on sound effects, but most data on capturing them comes from factory bulletins, standard references such as the Audio Cyclopedia, the Frayne and Wolfe test, and those on radio production. Though now unhappily no longer publishing, the fine magazine *Tape Recording* gave many hints, including a good series by L. I. Farkus.

The number of mikes available today as compared to the earlier days is fantastic, with a virtual "forest" growing from RCA, E-V, Sony/Superscope, Neumann, and others. However, even in the old days there was a good choice from Shure, Astatic, Brush, and E-V, not to mention the imports via the mail-order houses from Telefunken and others. One midwestern sound service engineer has made a hobby of collecting mikes from the 1920-1940 period and presently has more than 300 of them. The poorer designs from the period all too often had response patterns in which the horizontal and vertical fields did not match up, and substitution of a new ribbon for the original may make very little difference. The present-day ribbons, including those from such European firms as Beyer, are made pretty ruggedly and some models even have replacement ribbons which the user can install. The ribbon is still a basic mike, and in general offers smoother response than a dynamic with a mild treble rolloff and less trouble with "pops" and "spritz." The present-day dynamic has developed from a speech-only mike into the all-round super-cardioids, such as the E-V RE-20. The transition point—for the E-V line at least—would be the 650 "hammerhead," a literal grandchild of the 630-635 series and father to the 655C as well as the new cardioid series. The 650 has high output and almost flat response from 40 Hz to 15 kHz, while the 655 has a better bass, though not deeper, and treble response out to 20 kHz. Though a few complain of the E-V treble, I find only a slow rolloff at both ends relative to a good condenser mike.

Westrex introduced four dynamics from 1931 to 1939, and these include the 618A, the 630 "eight-ball," the 633 "saltshaker," and the classic 639 with

The ADC-XLM "...in a class by itself."



That's the way Stereo Review described our XLM. High Fidelity headlined their review, "Superb new pickup from ADC" and went on to say, "...must be counted among the state of the art contenders." And Audio echoed them with, "The ADC-XLM appears to be state of the art."

With the critics so lavish in their praise of the XLM, there's hardly any necessity to add anything. Far better to let the experts continue to speak for us.

Frequency response The CBS STR-100 test record showed less than ±1.5dB variation up to 20,000Hz. *Stereo Review*
...response is within ±2dB over the entire range. *Audio*
Frequency response is exceptionally flat. *High Fidelity*

Tracking This is the only cartridge we have seen that is really capable of tracking almost all stereo discs at 0.4 grams. *Stereo Review*
The XLM went through the usual torture test at 0.4 grams (some top models require more than a gram). *High Fidelity*
The XLM is capable of reproducing anything found on a phonograph record. *Audio*

Distortion Distortion readings...are almost without exception better than those for any other model we've tested. *High Fidelity*
The XLM has remarkably low distortion in comparison with others. *Audio*
At 0.6 grams the distortion was low (under 1.5 per cent). *Stereo Review*

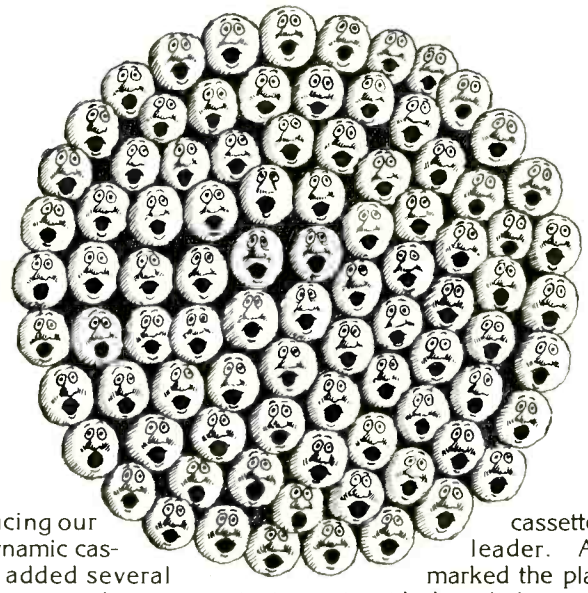
Hum and noise The XLM could be instrumental in lowering the input noise from the first stage of a modern transistor amplifier. *Audio*
The cartridge had very good shielding aga nst induced hum. *Stereo Review*

Price This would be a very hard cartridge to surpass at any price. *Stereo Review*
We found it impossible to attribute superior sound to costlier competing models. *High Fidelity*
Priced as it is, it is a real bargain in cartridges. *Audio*

The *High Definition* ADC-XLM $65.

AUDIO DYNAMICS CORPORATION
A BSR Company • New Milford, Conn. 06776

Maxell offers a huge improvement in cassette tape. Magnified 10,000x so you can see it.



Introducing our new Ultra Dynamic cassette. We've added several new improvements to the cassette. And a little more Ultra to the Dynamic.

Our smallest improvement is our biggest improvement.

We reduced the size of the tiny PX gamma ferric oxide particles on the surface of our tape. The Hz now go up to 22,000 Hz so you get even higher highs. The signal-to-noise ratio's now 8dB more than ordinary cassettes—which means you get less noise from your cassette player and cleaner, clearer sound from your cassette. And the dynamic range is wider so you can turn the sound up loud enough to disturb the neighbors without worrying about distortion.

Little pad finally gets grip on self.

Every cassette has a little pressure pad to keep the tape pressed firmly against the tape head. Other cassettes keep their pads in place with glue—or rather don't keep their pads in place with glue. So we've designed a little metal frame that holds the pad in a grip of steel. And now the tape can't push the little pad out of place—and you don't need to worry about signal fluctuations and loss of response any more.

An improvement you can see but can't hear.

The first five seconds of our new cassette is a timing leader. And we've marked the place where it starts with three little arrows so you'll always know exactly where you are.

Amazing new miracle ingredient fights dirt fast!!!

But the leader's also a head-cleaner and what's amazing, new and miraculous about it is that it doesn't rub as it scrubs as it cleans. Because it's non-abrasive. So it keeps your tape heads clean without wearing them down.

Our new long-playing cassette is shorter.

Our new UDC-46 is twenty-three minutes per side. Which very conveniently just happens to be the approximate playing time of your average long-playing record. (Our other cassettes are 60, 90 and 120.)

And that's our new improved Ultra Dynamic cassette.

And its ultra dynamic new improvements.

maxell.

The answer to all your tape needs.

Maxell Corporation of America, 130 West Commercial Avenue, Mobnachie, N.J. 07074

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ribbon and dynamic units. RCA offered several mikes in this period, the 88, a 50 series, the 44, and a 77 ribbon, and this 77 series were the first cardioids in movie use. Westrex and RCA also offered condenser mikes, but the first break came with the Cook-Capps low-cost units, followed in later years by the Stephens and the two Syncron mikes, these last being made by Cerwin-Vega. AKG and Sony have some fine condenser mikes, and I have a friend who likes the Altec 21-B introduced in 1950 for its wide range of response. Among the Neumanns, the U-47 is the quietest and the 201 the smoothest; another favorite is the U-67. AKG's 451E is also a good example of this type. Two key factors led to lowered costs of these mikes: One was the JFET, the other the electret principle, finally made practical.

In this brief space, it is difficult to give very many basic and firm rules, without immediately having to start listing exceptions. One thing I do recommend strongly is the use of headphones, since after you have worked with them for a while you will be able to "hear" the acoustics and sense the proper location for the mike from the nature of the sound to be recorded.

Once you've gotten your effect down on tape, there are some tricks you can do with speed changes and graphic equalizers. For example, a train or trolley will sound as though it is going into a tunnel if you increase the level at 320 Hz, and it will sound as if it is crossing a steel-beam bridge if you pulse the 2.5 kHz control. If you do both, the train will enter New York City and head toward Grand Central Station. It is also possible to make monsters of chickens by halving the speed and zebras of dogs by doubling the speed. I also had a tape of a storm at the seashore into which I added some variations at 640, 1280, and 2500 Hz that subjectively seemed to be rumble, whine and echo (sounds like some comedy team).

If you feel ambitious enough, you might want to transfer your effects to disc, using either a commercial service or—as this collector has done—with a home-brew "Rube Goldberg" cutter and table. Details on this will have to wait for another time, as they are outside the proper scope of this mikes and effects article. However, if you're intrigued by such a project but put off by the amount of work involved, let me tell you about Matthius Baldwin, a Philadelphia watchmaker, who in 1832 built a steam locomotive. The moral is, I guess, that there's still some hope for us amateurs.

NATURAL SOUND REPRODUCTION.

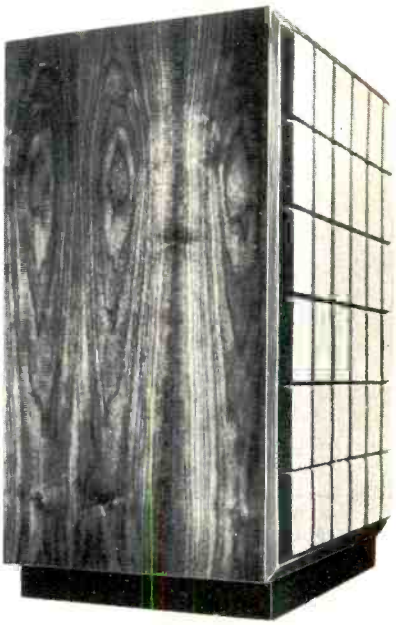
The Thrill of it all...

What's natural sound reproduction? It's sound so real you can see swaying palm trees and grass skirted dancers . . . feel the sun-warm sand beneath your feet . . . and smell the salt sea air . . . right in your own living room.

Making a speaker deliver sound this accurate takes years of experimenting and testing as well as solid engineering know-how. Here at UTAH, speakers are our only business. Our sound specialists never stop trying to find ways to improve the quality of our speakers. In fact, we design, engineer, and produce more speakers and speaker systems than any other manufacturer in the country.

When you play your favorite music through a UTAH speaker, you're hearing those clear, thrilling highs, deep, resonant bass notes, and smooth, mellow mid-range tones as closely as you can to the way they were originally recorded.

Natural sound reproduction. What else did you expect from the country's largest speaker manufacturer? The best . . . naturally. Write for the name of the UTAH dealer nearest you.



MP 3000.

This superb speaker system boasts 2 tweeters, 1 mid-range, and 1 woofer. Featuring separate controls for tweeters and mid-range, the MP-3000 handles 50 watts program material (100 watts peak). Frequency response: 20-20,000 Hz. Cabinet: 27" x 20½" x 14". Shipping weight: 70 lbs. **$199.95**

utah

UTAH ELECTRONICS

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New Loudspeaker With Extended Bass

William H. Watkins*

SOME OF THE MOST popular high-performance loudspeaker systems designed for the home utilize the acoustic-suspension or sealed-box principle. During the past several years, many improvements have been made in the woofer for this type of loudspeaker. New and improved materials and processes, such as polyurethane surrounds, vacuum-formed cones, and high-temperature voice coils, have contributed greatly to power-handling ability and smoother upper-range response. The performance below 500 Hz or so, however, has not been significantly improved, with the low bass response and distortion of the better units of some 20 years ago being essentially equal to those of present day design.

A problem with conventional design is that the laws of physics determine the maximum piston-band efficiency and low-frequency cutoff that can be obtained in a system with a given cabinet volume. If more efficiency is desired, bass response must be sacrificed, and conversely to obtain more bass, it is necessary to sacrifice efficiency.

One known method of circumventing this physical limitation is the negative-spring principle (1), which is somewhat complex in design. Another is the use of electronic equalization to modify the amplifier response, and this requires additional wattage for any gain achieved. The purpose of this article is to describe a third method that is both simple to implement and that requires no additional wattage from the amplifier.

Back EMF

Since the above problem is related to the motion-induced back EMF generated by the speaker at fundamental resonance, we need to understand its nature. The magnet-voice coil assembly of all dynamic speakers of the type we are discussing is a motor. An induced voltage is produced whenever an electrical conductor moves through a magnetic field; this effect is exhibited by all motors, whether rotary (the conventional type) or linear (such as a speaker) and is termed "back EMF" because its polarity is always opposite to that of the externally applied voltage. So this motion-induced back EMF opposes the amplifier voltage, raises the speaker's impedance, and reduces the current flow through the voice coil.

Since this back EMF is caused by the motion, i.e., velocity of the cone-voice coil, let us see how velocity and thereby back EMF, behaves. In a sealed-box, the enclosed volume of air is not a controlling or limiting factor above the region of resonance. Velocity is low, the air in the box is compressed and rarefied very little, and the air volume reactance is negligible. The output is determined by the cone area, motor, and mass of the cone and voice coil (2). Starting at 500 Hz, then, as we come down in frequency, the mass is "easier" to move due to decreasing mass reactance, and velocity doubles with each lower octave. In this area, and down to around 100 Hz, velocity is low, very little back EMF is generated to oppose the amplifier voltage, the back EMF is not a principal factor in determining current flow through the voice coil, and current flow is determined mainly by the d.c. resistance of the voice coil. This is the piston-band of operation where the decreasing air load resistance is offset by the increasing cone velocity, and the speaker output is approximately flat. As we move below 100 Hz or so and approach resonance, the air in the box enters the picture. The mass begins to resonate with the compliance of the air in the box, and velocity increases. When resonance is reached, the moving mass is at any instantaneous moment supplying energy to the air in the box (compressing or rarefying it) due to its kinetic energy, or the air in the box is supplying energy to the moving mass (forcing it in or out) due to its pressure or rarefaction. Thus, the mass and elasticity of the air in the box neutralize each other at resonance and velocity is greatest. With velocity greatest at resonance, the back EMF is greatest, and will raise the impedance by a factor of about 3 to 6. If the voice coil d.c. resistance is 4 ohms, the back EMF will add about 1 ohm at the resistive point in the 100 to 500 Hz range for a total of 5 ohms impedance in a typical woofer. Then below 100 Hz the increasing velocity may add 16 ohms at the resonant point due to the large back EMF, and the total impedance will be 20 ohms. This is shown in Fig. 1. Since the two points are resistive, we can calculate the power input to the voice coil by

$$\text{Watts} = \frac{E^2}{R}$$

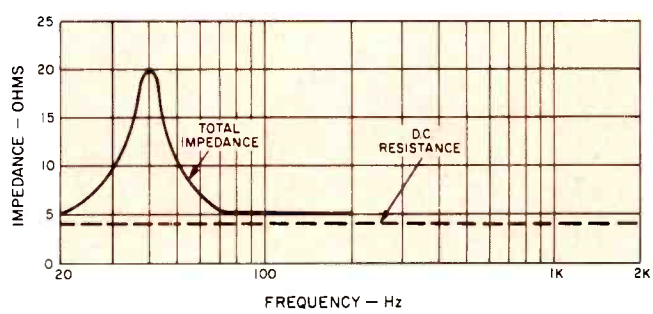

Fig. 1—Typical impedance curve. The total impedance is the d.c. resistance plus the reactance due to the back EMF.

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Where E = Voltage from the amplifier in volts
R = Resistance in ohms.

With E essentially constant, and with a voltage of 10 volts, the voice coil is absorbing 20 watts at the point above resonance and only 5 watts at resonance. Since a speaker is more efficient at resonance, this condition produces flat response in conventional design. It is clear however, that while the back EMF was not a principal factor in determining current flow and wattage through the voice coil above resonance, it has become the dominant factor at resonance and has severely limited the wattage intake of the speaker. In fact, if the power intake at resonance could be brought up to 20 watts and electro-mechanical losses were negligible, the relative bass output would increase by 4 times, that is, it would be elevated 6 dB above the higher-frequency output. If the output at these higher frequencies above resonance could then be increased, the net result would be an over-all higher output.

Dominant Parameters

While the device to be decribed may be characterized in terms of Q_T (shape of curve at resonance), it is believed that the approach used here offers a better "feel" for the principle involved. Although some approximations are necessary due to different factors in dominant control in given areas, the pattern of behavior is clearly shown (3).

The reasonable assumptions made are that the inductance of the voice coil below 500 Hz is negligible, that the amplifier source resistance is negligible (a high damping factor), and that good design is used throughout, and in particular in keeping proper control of the mechanical and acoustic resistive elements (4).

Since acoustic power output is proportional to velocity squared times the radiation resistance, and back EMF is also proportional to velocity, we will work mainly with this term. In the region above resonance and below 500 Hz, say at 200 Hz, where the cone of a 12-in. woofer is vibrating as a piston, the cone velocity is

$$V = \frac{F}{Zm} \qquad (1)$$

Where F = Force applied to the voice coil
Zm = Mechanical impedance

And F = Bli

Where B = Flux density in the air gap (2)
l = Length of voice coil wire in the magnetic field
i = Current in the voice coil

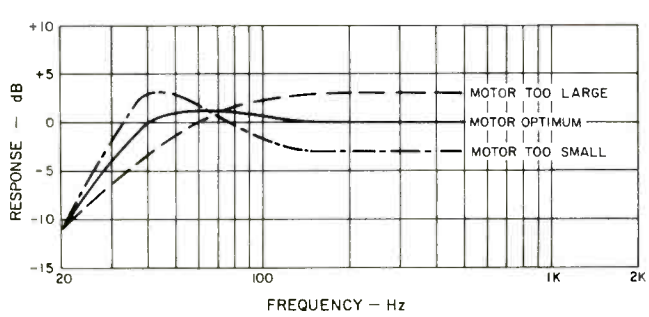


Fig. 2—Response for different values of motor strength in the region of fundamental resonance with conventional design.

So $$V = \frac{Bli}{Zm} \qquad (3)$$

To find the current, the total electrical impedance (5) is

$$Z_{ET} = Rc + \frac{B^2l^2}{Zm} \qquad (4)$$

Where Rc = Electrical impedance of the voice coil with no motion.

Now, above resonance (Fr), Rc dominates, and the current is

$$i \approx \frac{Eg}{Rc} \qquad (5)$$

Where Eg is the amplifier voltage.
Substituting for i in Eq. 3, velocity above resonance

$$V > Fr \approx \frac{Bl\,\frac{Eg}{Rc}}{Zm} \approx \frac{Bl\ Eg}{Rc\ Zm} \qquad (6)$$

Then at resonance in eq. 4, $\frac{B^2l^2}{Zm}$, the motional impedance term which is responsible for back EMF dominates, and the current is

$$i \approx \frac{Eg}{B^2l^2/Zm} \approx \frac{Eg\ Zm}{B^2l^2} \qquad (7)$$

Substituting for i in Eq. 3, velocity at resonance

$$V_{Fr} \approx \frac{Bl\left(\frac{Eg\ Zm}{B^2l^2}\right)}{Zm} \qquad (8)$$

And

$$V_{Fr} \approx \frac{Eg}{Bl} \qquad (9)$$

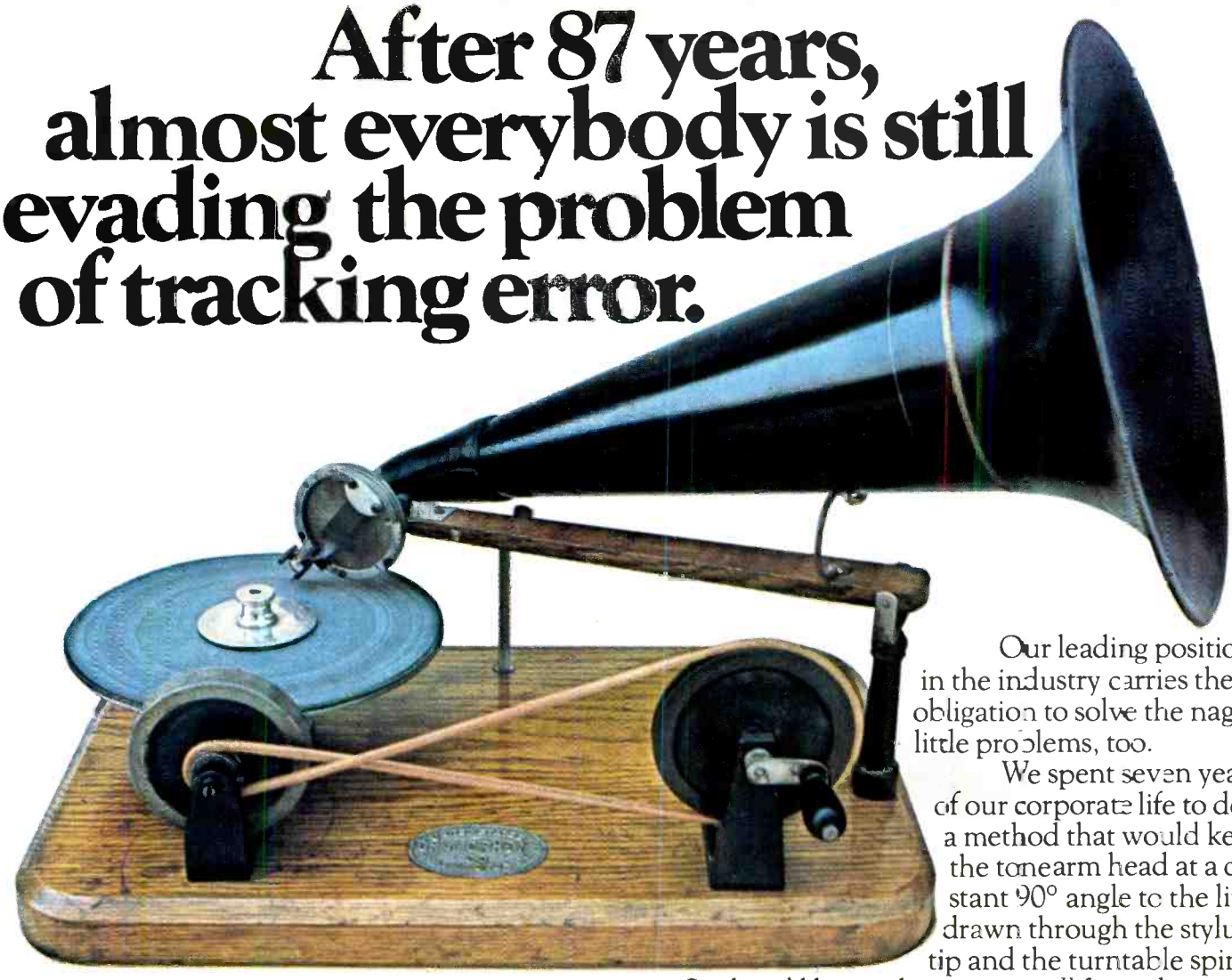
Zm varies with frequency, is a minimum at Fr, and eq. 4 shows that this low mechanical impedance at Fr is reflected back into the electrical circuit by the back EMF as a high impedance. So, as noted before, the voice coil does not absorb nearly as much power at Fr as it can above and below. Equation 9 shows that motor (Bl) is mainly in control of output at Fr. Since back EMF is proportional to Bl, a higher Bl gives more back EMF at resonance, and the velocity and acoustic output go down. Equations 6 and 9 show that for a given speaker, that is, a given area of cone, mass, and volume there will be an optimum value, neither too small or too large, of Bl product that will give the flattest response.

The See-Saw Effect

Now let's look at the performance of a loudspeaker system in which the sealed volume, cone area, and mass are kept constant and plot the results of changing motor strength. In Fig. 2, as shown by Eq. 6, in the 70 to 500 Hz range if Bl is increased output will increase, if it is decreased output will decrease. However, around resonance the reverse happens. As shown by eq. 9, increasing Bl decreases output and decreasing Bl increases output at resonance. The results shown in Fig. 2 are very similar to what would be obtained by denormalizing the standard Qt curves of .5 and 2 in Fig. 3 (6). The curves of Fig. 3 are generally referred to in discussing bass response. However, it must be noted that they have all been normalized with respect to actual flat band efficiency of each; i.e., the O dB point on the vertical axis is redefined for each speaker, regardless of actual efficiency. This normalization process does however, have a disadvantage, in that when adjusting the Bl product, it does not clearly show that a highly damped speaker will be more efficient above resonance than a less damped unit which peaks at resonance.

From Fig. 2 it can be seen that for a given speaker system, and where flat response is desired, the motor must be of

After 87 years, almost everybody is still evading the problem of tracking error.



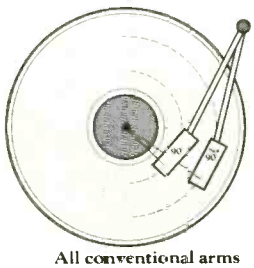
The tonearm of the original Emile Berliner gramophone of 1887 had tracking error.

Why? Because it was a pivoted arm with a fixed head, traveling across the record in an arc.

Just like *almost* every tonearm today.

The main difference is that today's tonearms don't have a big horn attached to them. But they have tracking error for the same inescapable reasons. The laws of geometry.

Mr. Berliner didn't talk about tracking error. He had bigger news for the world—and more important causes of distortion in his machine.



All conventional arms with a fixed offset have tracking error.

Today's manufacturers, on the other hand, have every reason to talk about tracking error, but they don't. They've refined their designs to the point where they call the tracking error and the resulting distortion "minimal," and they don't like to think about that irreducible minimum.

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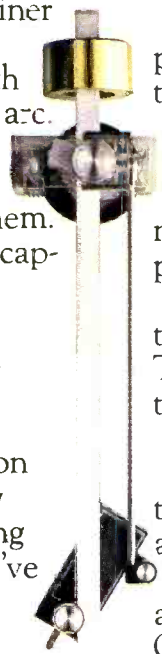

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We spent seven years of our corporate life to devise a method that would keep the tonearm head at a constant 90° angle to the line drawn through the stylus tip and the turntable spindle. So there'd be no distortion *at all* from that source.

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the proper strength. Balancing the bass response to the flat band is much like balancing a see-saw, with the pivot point at 70 Hz or so. If the motor is too small, efficiency is low, and there is a bump in the bass. If it is too large, efficiency is high, but the bass response is down. This also shows that purchasing the speaker system with the larger magnet (it is often assumed that the speaker with the largest magnet is best) could buy one a speaker with less than optimum bass response. By juggling motor then, there is an inevitable trade-off between bass response and efficiency in the flat band.

With a given size enclosure, the foregoing discussion shows the limitations which must prevail. Efficiency in the 70 to 500 Hz midrange area is tied to efficiency in the bass range, and the see-saw effect prevails. When driven by modern-day solid-state amplifiers with their high damping factors, measurements and/or reference to reliable test report curves will show that many of the better speaker systems will begin to fall off in bass response above resonance. This is due to the motor having been increased (the "see-saw" tilted up in favor of the midrange) in order to obtain reasonable efficiency, i.e., a total power output above resonance of about 80 dB SPL (sound pressure level), re

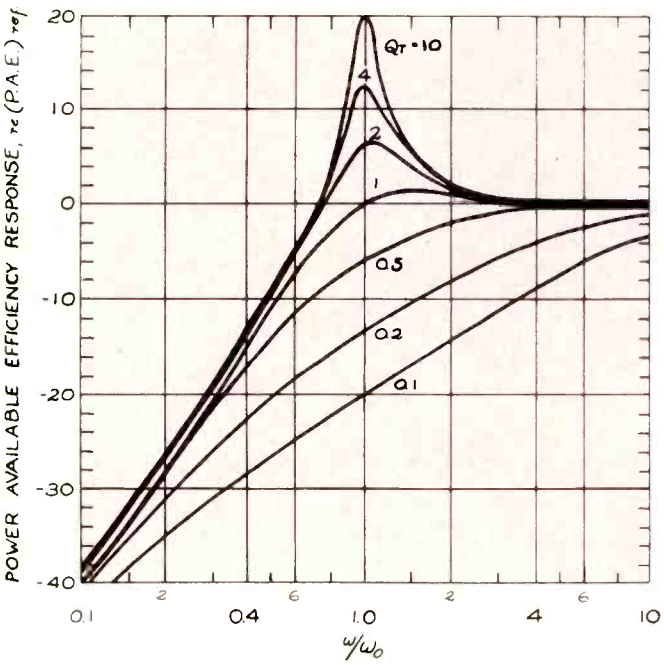

Fig. 3—Frequency response in the region of fundamental resonance for different values of Q_T.

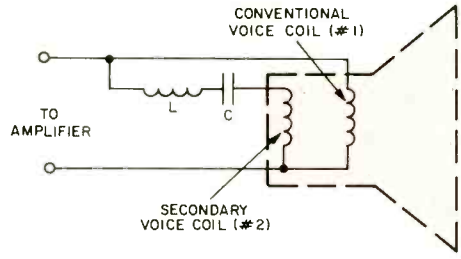

Fig. 4—Circuit for a dual-motor woofer. L and C are adjusted to resonate at the same frequency as the fundamental acoustic resonance of the woofer. The tuned bandpass circuit formed by L and C allow the secondary voice coil to operate only in the region of resonance.

to .0002 microbar at 1 meter distance for 1 watt of electrical input. So in the final analysis, the crux of the problem is due to the see-saw effect of varying motor strength. For good efficiency above resonance, we need a large motor; for good bass efficiency, we need a smaller motor.

Dual Motors

A solution is to provide the speaker with two motors. This would cut loose the see-saw effect, and we would be free to adjust upper range and bass separately, and without one affecting the other. The usual practice in adjusting motor strength is to vary the magnetic strength, that is, the B in the Bl product. To construct a speaker with two different magnetic field densities to drive the same cone would be both expensive and difficult to manufacture.

Let's look at the Bl product in a different way. In a conventional speaker, it is constant with frequency. Suppose we could in effect make l vary with frequency in such a manner that a lower value of Bl in one frequency range would not affect a higher Bl product in another range, and vice versa. Figure 4 shows a simple method of doing this. A second voice coil is wound over or under the conventional voice coil and is driven by a series LC resonant circuit adjusted to resonate at the same frequency as the fundamental mechanical-acoustical resonant frequency of the woofer. With proper design of the inductor L and capacitor C, the LC circuit presents almost zero impedance at resonance, and a sufficiently high impedance one octave either side of resonance to effectively remove voice coil 2 from the circuit. Thus l of voice coil 2 can be adjusted to eliminate the high value of motion-induced back EMF at fundamental resonance, a lower impedance path is provided at Fr for current flow, and bass response at Fr can be adjusted at will and independently of midrange response above Fr and response below Fr. Taking our example used before, this will allow the impedance to be lowered to say 5 ohms (compared to 20 ohms before) at resonance, and the wattage intake can be brought up if needed to equal that above resonance (20 watts). At this point it should be noted that no attempt is made to violate the laws of physics. A second motor is added that generates less back EMF and offers a lower impedance to the amplifier, allowing more wattage intake to the speaker in the region of a resonance, and providing more bass response. We are simply using the wattage in a given amplifier that was *there all the time*, but not being used in the bass range.

The additional cost of building a woofer of this type over a conventional woofer is a pleasant surprise. A voice coil gap of twice the width is not required (a wider gap requires a larger magnet to obtain the same flux density) to accommodate the extra voice coil. The outer and inner clearance spaces are the same as usual, and since a single layer coil with a notch in the top plate for the return end of the coil has proved satisfactory, the gap width need only be increased by about 25 per cent. Any extra money put into a larger magnet can be utilized to provide benefits that can be put to good use.

Once it is realized that the motor can be increased to the point of gap saturation with the attendant overdamping of bass compensated for by the second voice coil, the dividends of the design are apparent (as the flux density is increased, the impedance at resonance is increased, and fewer turns may be used in the second voice coil without its impedance falling to an unacceptable value). More overall efficiency, a smaller enclosure, extended bass as noted, or a combination of these may be achieved, according to how the parameters are chosen, and with the usual tradeoffs in these areas minimized. For the same shape of curve and volume of box the maximum realizable efficiency gain is calculated to be slightly over 3 dB.

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Shown above: (left to right) new walnut and white 901 Continentals; ebony, white and walnut 901s; (center) 501. Pedestals optional.

Several variations of the principle involved were tried, but more research is needed in some areas. While using the circuit of Fig. 4, if the main voice coil is disconnected at resonance, a further increase in acoustic output occurs. The measured EMF across it is higher than the amplifier drive voltage by about the same ratio as the ratio of turns of wire in the two voice coils. Inserting a parallel LC circuit in series with the main voice coil allows the increase to be utilized.* The Q of the LC circuits are controlled in a straightforward manner by the LC ratio. Mutual coupling of the two inductors in the two LC circuits can also be accomplished.

Performance

Figure 5 shows the frequency response of a woofer in which a gain of 5½ dB at 30 Hz was achieved. In all the tests the speaker was sunk in the ground with the front edge flush with the ground surface, and well away from any buildings (180-deg. solid angle, free-field conditions), and the microphone was at 1-meter distance, on-axis with the geometric center of the speaker.

The following test equipment was used for the performance figures shown:

Hewlett-Packard 200-CD audio oscillator, Hewlett-Packard 400-H A.C. vacuum-tube voltmeter, Hewlett-Packard 120-B oscilloscope, Hewlett-Packard 330-B distortion analyzer, Crown DC-300A amplifier, B & W 410 a.c. vacuum-tube voltmeter & distortion analyzer, General Radio 1390-B random noise generator with p2 pink noise filter, General Radio 1564-A sound and vibration analyzer, General Radio 1962 ½-inch condenser microphone, General Radio 1560-P42 microphone preamplifier, General Radio 1562-A sound level calibrator, custom-built a.c. impedance bridge, and Eico 377 square wave generator.

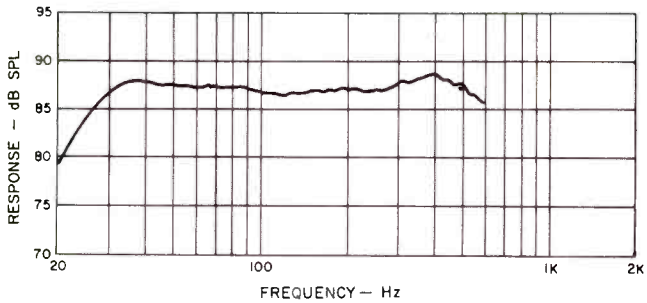

Fig. 5—Frequency response of a dual-motor (1 watt into rated impedance of 4 ohms), mike at 1 meter on-axis, 180-deg. radiation angle, and free-field conditions. SPL ref. is 0.0002 μ Bar.

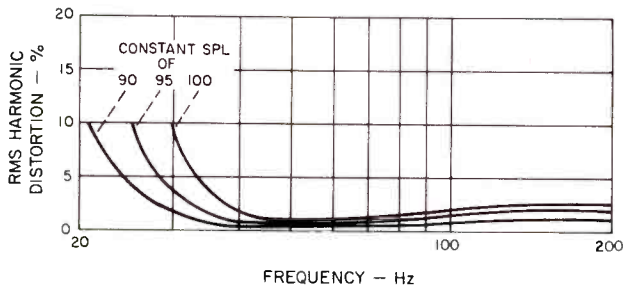

Fig. 6—Total rms harmonic distortion with free-field conditions as noted previously. The sound pressure level was held constant for each curve, and the frequency lowered until the distortion reached 10 percent.

The woofer is a 12-in. unit (9½-in. actual cone diameter), mounted in a box of two-cubic-foot net internal volume, and the box filled with a suitable amount of fiberglass.

In constructing this unit, primary emphasis was placed on extending the bass, slight emphasis on efficiency, and none on reducing enclosure volume. The reason for this is that the writers experience has shown that, other things being equal, the loudspeaker listener will usually prefer the speaker with the cleanest and most extended bass response.

Taking 87 dB SPL (reference .0002 microbar) produced with 1 watt input (2 volts into a rated impedance of 4 ohms) as reference, the response is flat to 30 Hz, and down only 3 dB at 25 Hz. The response is flat ± 1½ dB from 27 thru 600 Hz. It has been stated (7) that a range of 30 to 15,000 Hz is required to reproduce orchestral music with no discernible frequency discrimination. This is in agreement with the lowest musical tones generally found in recorded program material of the piano, double-bass viol, harp, organ, and drums. The curve of Fig. 5 is unusual, considering the 2-cubic-foot volume, the overall efficiency, it being flat to 30 Hz, and that no amplifier equalization is used.

Total rms harmonic distortion is shown in Fig. 6. Note that the curves are for constant sound pressure levels, not constant wattage input. A specification showing low distortion down to 30 Hz with a constant wattage input is not very meaningful, if the acoustic output of the speaker is down 10 dB at this frequency.

Free-field specifications can be taken as statements of fact and offer data for direct comparison, but an assessment of performance in the listening room—even an "average" room—involves many variables, and becomes a difficult task. Furthermore, no standard has been adopted on either the maximum SPL needed or the limit on distortion. These are subjective matters, and it is therefore difficult to assign rigorous figures. The effect of boundary reflections on the response in the listening area is also of importance. Much work is now being done in these areas, and we can expect more useful information to become available.

Nevertheless, it is desirable to assess the performance of a new device in terms of practical use, and an attempt will be made to present the low bass capability in a manner that will allow the reader to make his own evaluation. We need to decide on the lowest frequency to be reproduced, the SPL it is to be reproduced at, and the amount of distortion to be tolerated. Let's set the low frequency at 30 Hz as mentioned earlier, the SPL at 100 dB (extended listening at levels much higher than this can cause hearing impairment), and the distortion limit on the order of 5 percent (this level of distortion is tolerated in loudspeakers designed for demanding use at higher frequencies and lower volume levels than we are talking about here). At 30 Hz and radiating into 180 deg. free-field, the distortion at 97 dB SPL at one meter is 5.2 percent. Now if we move the speaker into the listening room, place it against a wall and add another unit for stereo use, the SPL will double and will be 100 dB. In an "average" room of 2500 cubic feet, and at this low frequency, the reverberant field SPL should equal the direct field SPL at about one meter (8), so the 100 dB SPL should prevail in the listening area. On this basis, we have a woofer in a two-cubic-foot box, flat to 30 Hz, with a stereo pair producing 100 dB SPL at 30 Hz in the listening area at a distortion level of only 5.2 per cent, and with a power input to each of only 10 watts. If the SPL is lowered to 95 dB, the harmonic distortion drops to 2.3 percent, and at 90 dB it is 1 percent. This 5.2 percent low-bass harmonic distortion in a woofer still allows for a surprisingly clean waveform, if clipping of the peaks and sharp aberrations are missing. Figure 7 is an oscilloscope

*A capacitor in series with the main voice coil was used in the test speaker for this article.

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photo of the acoustic output at 30 Hz at the SPL producing 5.2 percent distortion.

The impedance curve for a complete system is shown in Fig. 8. In one prototype, the curve was ruler flat below 200 Hz, however, the best overall results were obtained with some deviation from this ideal, and the curve shown is the one obtained from the design giving the other data presented in this article. The minimum is 4½ ohms at 25 Hz, the maximum is 7 ohms at about 2200 Hz, and a 4-ohm rating is appropriate. It does not fall under 5 ohms below 20 Hz, and above 20,000 Hz, there is a gradual rise due to the usual inductance of the tweeter voice coil. The conventional bump at bass resonance is missing, due to the effect of the second voice coil increasing the power intake in this region.

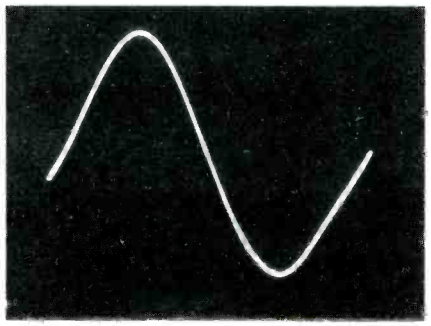

Fig. 7—Acoustic output of the woofer at 30 Hz and 97 dB SPL. Free-field conditions as noted previously. (Oscilloscope of microphone output).

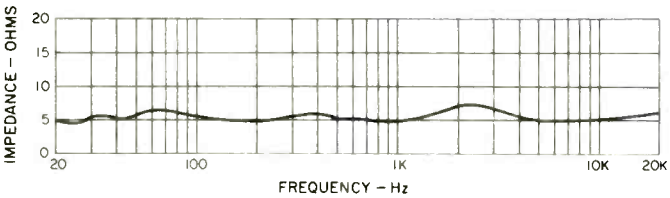

Fig. 8—Impedance of a system using dual-motor woofer.

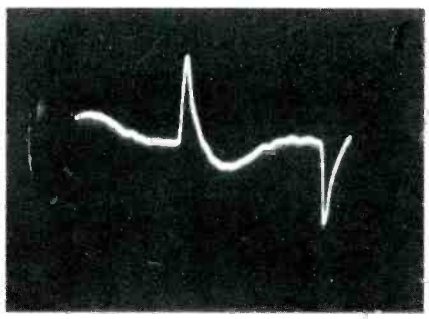

Fig. 9—Acoustic output of the woofer in response to the step-front of a low-frequency square wave.

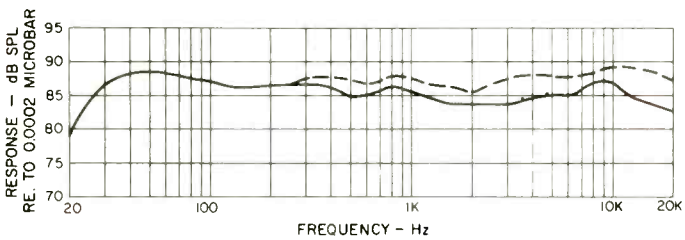

Fig. 10—Frequency response of a three-way system using the dual-motor woofer with one-third-octave random noise. Speaker radiating into 180-deg. free-field conditions as noted, with one-watt input.

Transient response refers to the ability of the woofer to follow the starting and stopping (attack and decay) of sudden changes in the electrical signal. In a woofer, this is a function of the system Q. It is generally considered that a Q of one offers the flattest response with a minimum of ringing, and the response curve of Fig. 5 conforms to this shape. Since the woofer is driven by a series resonant LC circuit, we need to see if the transient response follows the rules. It has been shown (9) that a woofer in a sealed-box with a Q of one will have about 33 percent overshoot to a step-signal, and will decay within one cycle. Figure 9 is an oscilliscope photo of the wave-form of the acoustic output in response to the step-front of low-frequency square wave. The overshoot past the neutral or start position is exactly that expected with a Q of one, and shows that the woofer is properly damped both acoustically and electrically. This test was made indoors, and the small ripples in the waveform are due to room boundary reflections.

System Performance

The frequency response of a complete system consisting of a 12-in. woofer, 4½-in. mid-range, and 1-in. dome tweeter is shown in Fig. 10. One-third-octave bands of random noise were used to produce the curves. The dot-dash line is the on-axis response, and the solid line is the average of five frontal curves; one on-axis, and four at 45 degrees, with the microphone at the sides and ends, one meter away from and aimed at the geometric center of the box. Taking 85½ dB SPL as reference, the lower curve (a good indication of the overall response), shows the system to be ± 3 dB from 23 through 20,000 Hz. The system's impedance curve is that of Fig. 8. Although the output of the mid-range driver needs to be increased slightly, the response is relatively smooth.

In listening comparison using program material with deep bass content, the low bass is simply "there." The lowest notes are reproduced cleanly at equal volume, and the temptation to turn up the bass control is missing. The low notes have an "easy unstrained" sound, which is attributed to the linearity and low phase shift of the load presented to the amplifier.

In summary, the design does not involve trade-offs in areas of performance, requires no additional amplifier power or equalizer, has the advantage of simplicity of construction, and offers an improvement that is readily discernible on listening.

A patent application has been allowed for the loudspeaker system described in this article.

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Part XII

Martin Clifford

The signal that appears at the output of the demodulator or a first audio amplifier in either a tuner or receiver is like a new born babe—in no position to be disputatious, but nevertheless full of promise. At this point in the electronic circuitry, the audio signal has left the womb of its surrounding carrier and now is naked and unsupported. What it needs is substantial nourishment in the form of voltage and current to enable it to grow to cope with the realities of an outside and demanding world. In the case of the audio signal, that outside world is a speaker system.

There are two basic components that will help the audio signal achieve maturity. One of these is the pre-amplifier, more familiarly known as the pre-amp, while the other is the main or power amplifier. At its entrance to the pre-amp, the signal is still in its electrical form, whether the signal comes from the tuner, a cartridge, or a tape unit, and so can be modified by all the electronic circuits we have at our disposal. In the case of the tuner, it would be nice to assume that the signal at this stage is a perfect replica of the original audio modulating signal used in the broadcast studio. However, in the process of getting to your tuner, the signal was forced to undergo a rather horrendous experience. At the station, it was converted from sound energy to electrical energy with the help of a microphone, unceremoniously loaded onto a carrier wave, and then literally thrown out to fend for itself in space. In its travels it was unmercifully harassed by noise signals, and all other kinds of signals as well, finally arriving emaciated and almost non-existent at your receiving antenna. Here it was picked up, given an invigorating dose of amplification, cleansed of miscellaneous unwanted voltages that clung to it like so many burrs, finally arriving at the output of the tuner. At this juncture it is strong enough to produce sound in a pair of headphones, but that is about all. At this point also the signal should be a faithful replica of the original energy conversion process—that is, the change of sound energy into electrical energy by the microphone. Whether it really is depends on more than just the electronic ability of your tuner or receiver. In FM broadcasting, high fidelity starts at the broadcasting station—or it should. It is entirely possible that your hi fi components might be superior, electronically, to those at the station, since such is the advanced state of the art in the hi-fi components industry. And you may also not know whether the signal followed a direct path from the station to your receiving antenna, or used a multipath, a situation in which the signal competes with itself—the direct signal and that portion that is bounced off buildings or other structures. And so the assumption of an FM signal input to a pre-amp is high-fidelity isn't always justified.

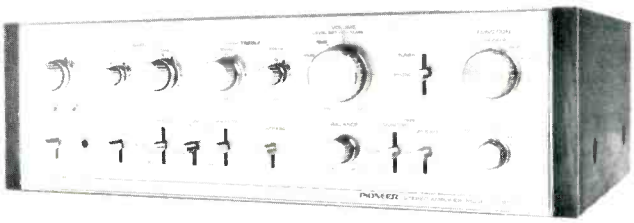

Fig. 1—Integrated stereo amplifier.

Sound Sources

The FM signal is just one of several that can be fed into the pre-amp. You might opt for records, or for signals supplied by a cassette deck, a tape deck or a microphone. All of these sound sources require additional amplification—amplification that can be supplied by the pre- and main amps. Note the economy of this arrangement. The same amps can be used for a number of different sound sources.

Terminology

The pre-amp and power amp can exist as separate components. The pre-amp then becomes the intermediary link between the tuner and the main amp. An integrated amp (Fig. 1) consists of a combination of pre- and main amplifiers in a single enclosure. The advantage is that the pre-amp does not require any connecting cable to the main amp. There is also some economy since both units are housed in one box. However, this joining of the pre- and main amplifiers is at the expense of component flexibility, and so some integrated units have a provision to let the owner use either the pre- or main amps separately, or as an integrated unit. This gives you the best of both worlds. Of course, if you have a receiver instead of a tuner, the problem of separated or integrated amps does not arise, since the receiver is equipped with both amplifiers. Originally, amplifiers used in hi-fi receivers were rather low-power types, making separated or integrated amps preferable. However, modern receivers now have amps whose power ratings are substantial. The disadvantage, of course, is that if you want to modify or upgrade a system that has a receiver, then, in effect, you must dispose of a tuner cum pre-amp cum amp, for that is what a receiver is.

The receiver with its built-in amps has powerful arguments in its favor. One of these is economy. A combined component such as a receiver costs less to manufacture. The receiver needs less room than a tuner-pre-amp-power-amp combination. And, finally, there is no need to make connections. But in the face of all these considerations, the separate tuner, separate pre-amp, separate power amp (or integrated amp that has provision for operating the pre- and main amps separately) supplies flexibility. It facilitates moving up to a multiple-amp system or to a quadraphonic setup without sacrificing equipment. Individual components also supply the satisfaction of having a custom-built, rather than an "off the rack" hi-fi system.

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STEREO REVIEW, *America, July 1973*

"I can't say whether or not ESS speakers are the loudspeakers of the century, but the innovation which they claim to have made in the field of sound is real and authentic . . . The amt 1 is reserved for perfectionists, for persons who are never satisfied, for those who, in search of musical fidelity, have listened to all the great loudspeakers of the world. They are superior both to dynamic and electrostatic speakers for their unmatched definition of timbre and great dynamic capability."

STEREOPLAY, *Italy, May 1974*

". . . After listening to the amt 1 for a period of time, there was a great temptation to ignore the measurements, good though they are, and simply assert that it is a superb speaker . . . This is the kind of performance that is extremely difficult to predict from the standard test results. Great clarity, coupled with a wide, even response and no coloration is the almost unbeatable combination that puts the amt 1 right up front."

AUDIO SCENE, *Canada, November 1973*

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SON MAGAZINE, *France, October 1973*

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ELECTRONICS TODAY INTERNATIONAL, *Australia, August 1974*

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Position of the Amp

The position of the amplifier with reference to the other components in the hi-fi system depends on whether we are talking of a receiver or a tuner. Figure 2 shows a receiver (A). Since both pre- and main amps are contained within the receiver, the question of position does not arise. Connect an FM antenna to the receiver "ant" terminals, a pair of speakers to the speaker terminals, and the hi-fi system is ready to go. With a tuner, though, we have more choices: an integrated amplifier (B) or separate pre- and main amps (C).

These are all fundamental hi-fi systems and do not take advantage of the fact that other input signal sources can also be used. Figure 3 shows a sophisticated arrangement using a pair of record players, two tape decks (one of which could be a cassette type and the other open reel), stereo headphones which can plug into the front of the integrated amp, one or more microphones, and a pair of speaker systems.

Still another arrangement, known as a multi-amp system, is shown in Fig. 4. In this setup, there is one pre-amp contained within the integrated amplifier. The integrated amplifier also has a main amp, a power amp that is used solely for the bass tones. There are two additional power amps, both driven by the pre-amp—that is, they receive their input signal voltage from the one pre-amp in the integrated amplifier. One of these power amplifiers handles all the mid-range tones while the remaining power amp takes care of all the high range. An electronic crossover network is used to supply each of the power amps with signals in the mid- and high ranges.

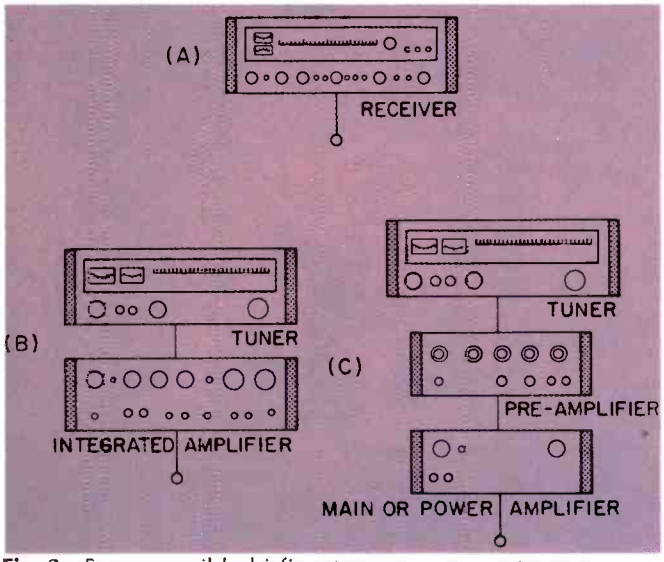


Fig. 2—Some possible hi-fi system arrangements.

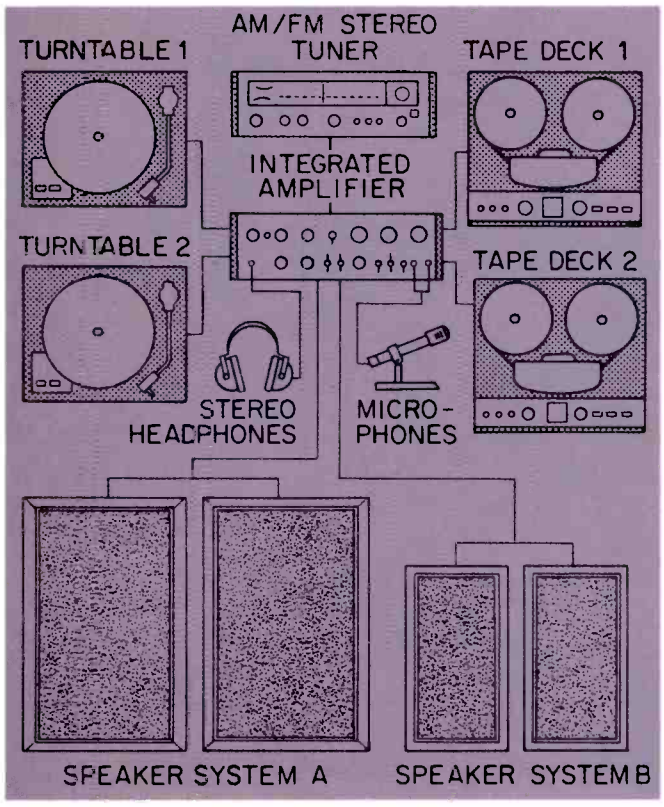


Fig. 3—A pre-amp or integrated amp can act as the control center for a number of hi-fi components.

Some Basic Differences

The pre-amp and the main amp are both audio amplifiers. The basic difference lies in their responsibilities. The pre-amp is a voltage amplifier; its function is to strengthen the signal voltage, to increase its amplitude without modifying the input signal waveform in any way. The main amplifier is also a voltage amplifier but since the speakers will be connected to its output, must deliver current in addition to voltage. But a combination of voltage and current means power. With this as its primary function, the main amp is sometimes called a power amp. The pre-amp

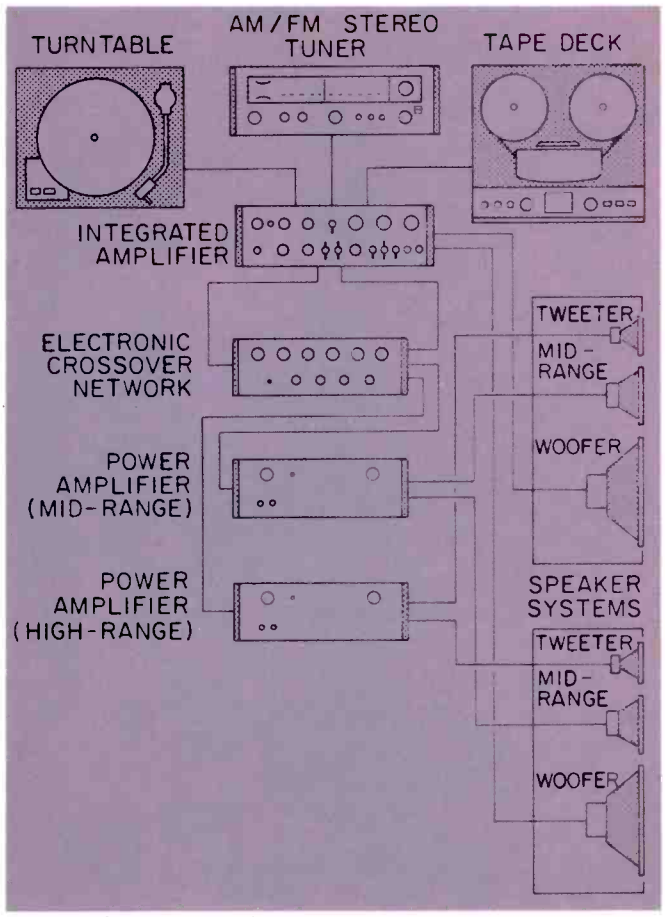
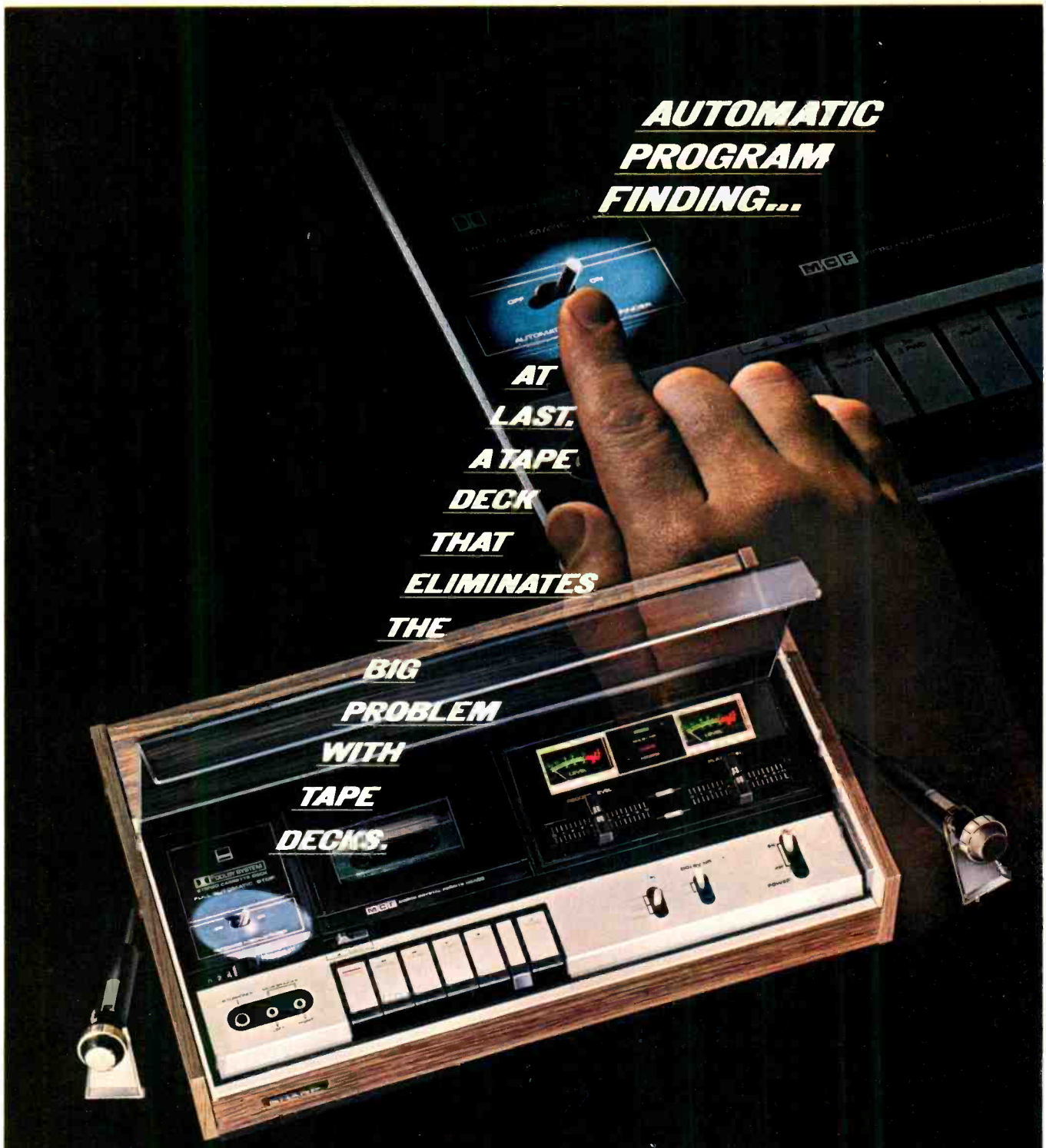


Fig. 4—The integrated amp can serve in a multi-amp installation more easily when its pre and main amp sections can be worked independently.



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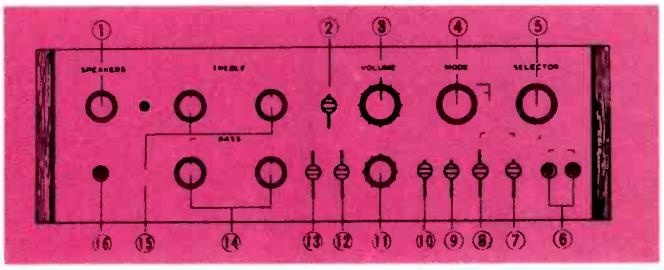
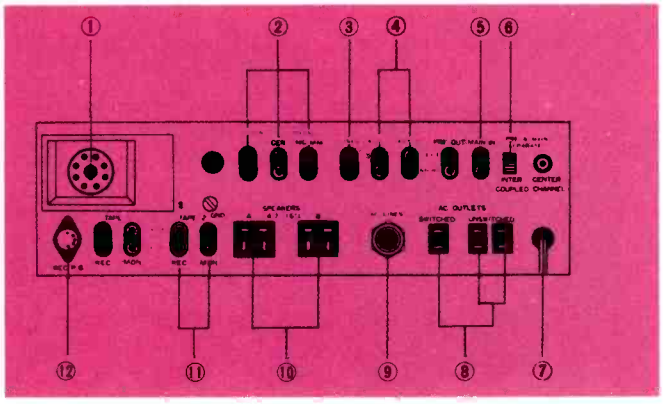
must strengthen the audio signal to the point where the output amplitude is strong enough to satisfy the signal input voltage requirements of the main amp. In turn, the main amp must furnish enough signal power—that is, both voltage *and* current—to operate the speaker system (or systems) to your listening satisfaction. Thus, the output of a pre-amp is measured in volts; that of a power amp in watts. The voltage output of a pre-amp, since it is a signal voltage, will be a.c. Its amplitude will be somewhere in the region of 1 to 5 volts, depending on its overall amplification. Some pre-amps do deliver more voltage than this; others less.



① POWER/SPEAKER SWITCH
② VOLUME CONTROL
③ MUTING SWITCH
④ MODE SWITCH
⑤ SELECTOR SWITCH
⑥ MIC JACKS
⑦ TAPE DUPLICATE SWITCH
⑧ TAPE MONITOR SWITCH
⑨ PHONO 2 MM/MC SWITCH
⑩ LOUDNESS SWITCH
⑪ BALANCE CONTROL
⑫ HIGH FILTER SWITCH
⑬ LOW FILTER SWITCH
⑭ BASS CONTROLS
⑮ TREBLE CONTROLS
⑯ PHONES JACK

Fig. 5—Front panel controls of an integrated amp.



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⑥ PRE & MAIN SWITCH
⑦ PLUG
⑧ AC OUTLET
⑨ FUSE
⑩ SPEAKER SYSTEMS
⑪ TAPE DECK
⑫ TAPE REC/P.B. CONNECTOR

Fig. 6—Rear panel connections for an integrated amp.

Sometimes a manufacturer's spec sheets will indicate two different voltages for the pre-amp's output. One will be rated output; the other maximum. A pre-amp might supply 3 volts rated output; 5 volts maximum.

Input Sensitivity

The amount of input is sometimes designated as the pre-amp's input sensitivity and in manufacturer's spec sheets is given in terms of voltage at a specific frequency, quite often 1 kHz. A turner, for example, might supply 200 millivolts to the pre-amp's input to produce 3 volts rated output. However, even though the pre- and main amps can be, and sometimes are, separate entities, they are related in the sense they must function as a team.

The Pre-amp as a Control Center

The pre-amp does more than get the signal ready for introduction to the power amp for it works as a control center for the hi-fi system whether it exists alone as an individual component, as a combined component with a power amp, or set up as an integrated amp. If a receiver is used instead of a tuner, then the only difference is that the pre- and main amps are contained within the receiver, and the concept of the component as a control center is still applicable. And so, front panel controls on a receiver, or on a pre-amp, or on an integrated amp (Fig. 5) can include bass and treble controls, a loudness switch, volume control, speaker balance control, a mode selector switch, 'phone and microphone jacks, tape duplicate and tape monitor switches, various noise filters, a speaker selector control, etc. Naturally, not all pre-amps are so thoroughly equipped since the degree of sophistication of the pre-amp depends fairly directly on what you are willing to pay.

The Rear Panel

Modern hi-fi components such as receivers and amps are distinguished by the fact that the rear apron or rear panel (Fig. 6) is as busy an area as the front. Again, what you have on the rear panel depends on the degree of sophistication you want. Here you will find convenience a.c. outlets, since the average home is simply not equipped with enough outlets for a multi-component hi-fi system. Some of the outlets are unswitched, meaning that any component plugged into such outlets must be provided with an on-off switch. You will also find one or more 'switched' outlets—those which are activated by the power on-off switch on the front panel of the amp. These outlets have power limitations—that is, they are limited in the amount of a.c. line power they can safely handle. 100 watts maximum is about what you can expect from either switched or unswitched outlets.

On the rear panel you will also find input connections for one or more turntables, the tuner, one or more tape decks, cassette deck and possibly a cartridge deck as well, and inputs for one or more stereo speaker systems. A well-designed pre-amp will have different phono inputs—magnetic or ceramic.

Program Selector

The program selector, sometimes called the function selector, chooses the program source that is to be fed to the input of the pre-amp. These inputs can be one or more mikes, phono, tuner or tape units such as cassette or open reel. There may also be one or more auxiliary (labeled "aux") inputs for other audio signal producing components.

Tape Monitoring

Tape sound can be switched on and off with the tape monitor switch, a useful control when making a recording on a cassette or open reel deck. If the deck has independent

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recording and playback heads—that is, if it is a 3-head deck—this switch supplies a double method of monitoring recording quality. When this control is in its "source" position, the original "before it gets on tape" sound is heard from the speakers; in the "play" position, the sound is that recorded on tape. By switching back and forth between the two switch positions, the source sound can be compared with the tape sound supplying a clue as to the proper setting of the "record level" control on the cassette or open reel deck.

Tape Duplicating

You can use the amp to duplicate tapes. If you tape record FM broadcasts, for example, it is difficult to eliminate unwanted sounds, such as an announcer's comments or advertising break-in. The tape can be edited by cutting and splicing, but an easier method is to use the technique shown in Fig. 7. This requires a pair of tape deck input terminals on the rear apron of the amp. First, tape a complete stereo FM program, including commercials and announcements, and then later re-record with another tape deck, wiping out the unwanted material.

Volume, Balance and Tone Controls

These controls on the amp front panel are rather obvious. The balance control, for example, when turned to the right does not increase right channel sound since it cannot do so. Instead, it reduces left channel volume and so the right channel sounds louder. Also, the volume and balance controls do not affect the signal supplied to the tape deck for recording. This does not mean you have no bass or treble control on tape, for the controls are effective during tape playback through the pre-amp.

Depending on the amp, some volume controls have a greater or lesser amount of electronic sophistication. Thus, some volume controls are assisted by a level-set control. The level-set control could be a click-stop device reducing amplifier output by a predetermined amount, such as 15 dB or 30 dB. The level-set control can then be regarded as a "coarse" setting, and the volume control as a fine or vernier adjustment. When a powerful integrated amp is used for listening at quiet sound levels, the volume control must be turned down almost to its minimum point, making fine adjustments for low volume listening levels sometimes difficult. However, the level-set control can push the volume down to the required level, thus letting the volume control have a greater arc of swing. This makes the volume control setting much less critical.

Another advantage of a level-set control combined with a volume control is that limiting power output also protects speakers, especially those of small input power handling ability from accidental damage that might occur when you turn up the volume by mistake. Using speakers having small power-handling capacity with a relatively high-power integrated amplifier may seem anomalous, but the situation can exist. With speakers of small power-handling capabilities it is advisable to have the level-set control positioned at -15 dB or -30 dB at all times.

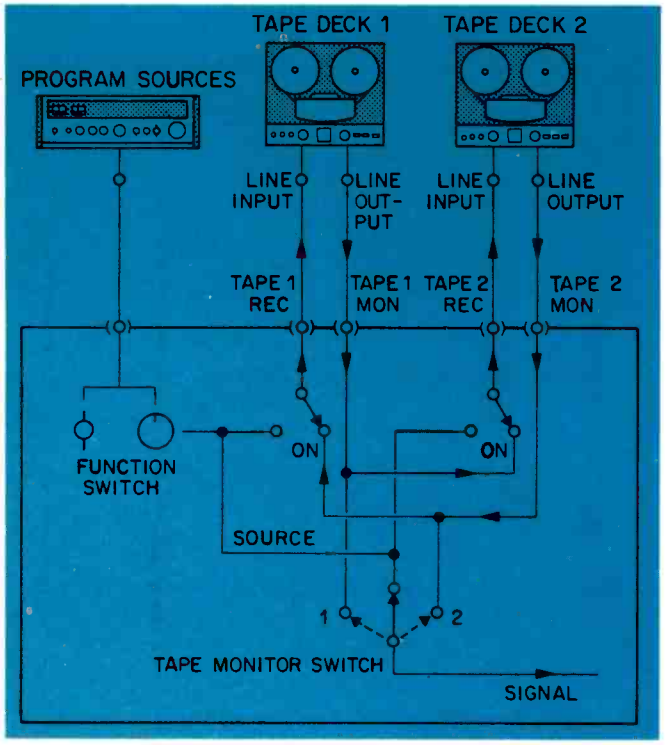

Fig. 7—Technique for tape duplicating.

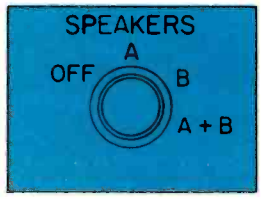

Fig. 8—Amp may be equipped with a speaker switch.

Frequency Response

The frequency response of an amp is the range of audio signals the unit will reproduce when an audio voltage is connected to its input terminals to yield 1 watt of output signal power. By itself, frequency response is meaningless unless some information is supplied about the amount of deviation, plus and minus, generally given in dB. A spot frequency can be selected as the reference. Thus, we might consider the output level of 1 watt at 1 kHz as indicative of 0 dB. The smaller the deviation in dB from the reference, the better.

An amplifier might have several frequency response figures. The same amp could be described as having a frequency response of 20 Hz to 18 kHz, ± 1 dB, or 18 Hz to 25 KHz, ± 3 dB. The deviation in dB is an indication of how far the amp wanders from fidelity.

Speaker Selection

Another highly useful, but possibly seldom used control on the front panel of the amp is the speaker switch (Fig. 8) often combined with the power on-off control. Aside from the power off position, this control could have three speaker selection settings. If you designate your left speaker as A and the right speaker as B, then this control can let sound come through the A speaker only, or the B speaker, or a combined output, that is, both A and B speakers functioning. There may also be a "speaker off" position, helpful when listening through headphones and you want the speakers silenced.

On to the Power Amp

The pre-amp in its dual role of amplifier and control center is one of the most important links in the hi-fi system. We've been following the audio signal since its inception at the source and so the speaker system is our logical destination. To get there we must go through the main or power amp—and that's exactly what we will do in the next installment.