Audio hi fi handbook

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VOLUME 1

a complete reference guide to all hi fi and stereo equipment including prices, specifications and comparisons

Audio hi fi handbook

VOLUME 1

Editor,
Eugene Pitts III

Publisher,
Jay L. Butler

CONTENTS

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134 N. 13th Street,
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Features 12 Darlington transistors in series output, 400 watts RMS stereo, amplifier and loudspeaker relay protection circuit, full complementary output stage, UNI-SINK heat dissipation system, direct power reading meters.

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Audio

Dear Audio Reader:

The Audio Hi-Fi Handbook is a unique publication in several ways. First, we've compiled a double-handful of the best articles published during the past few years. These pieces, written by industry leaders, reflect the latest thinking on every audio topic and contain information for both the beginner and the advanced audiophile.

Second, the Directory sections present specifications for more than 1,100 audio components in a convenient tabular form. You'll find it easy to compare specs for every top model. (Note that the data are supplied by the maker; they are not the result of our tests.)

To make the Directory easier to use, we've adopted some letter codes. For example , (B) with an amplifier means the unit is a basic amp; (K) indicates kit; (D) shows the unit is equipped with Dolby B circuitry. You'll find others explained with the various charts. Rather than use the substantially higher music power or peak power ratings, we list amplifier power in rms or continuous watts per channel at 8 ohms, with all channels driven (two for stereo, four for quadraphonic).

Also included in the Audio Hi-Fi Handbook is a listing of manufacturers' names and addresses so that you can easily write for additional information on the products listed, names of local dealers or data on products not included because of space limitations.

Sincerely,

Eugene Pitts

Eugene Pitts
Editor

PRODUCT PREVIEW DIRECTORY

Every year since 1958 Audio has presented a Product Preview Directory, giving specifications and photos for as many high fidelity products as is possible with the space allotted. In 1965, the tabular presentation was adopted in order to facilitate comparisons of the various figures. It should be noted that these figures are manufacturer's specifications, rather than the results of our tests, since it is impossible to test the more than 1000 products listed in such a short time span. However, it has been our experience that performance is generally as specified, though methods of test and measurement do differ from maker to maker.

For purposes of clarity, letter codes are employed in some sections, usually to show the speeds at which a unit will operate. Other uses of letter codes are (B) for basic amp with the model number; (K) with a price for kit, and (D) for Dolbyized with a recorder's price. In addition, we have tried to note under the model number when a unit is four channel, listing the type of system used in the Special Features column.

Amplifier power ratings are again listed in rms or continuous watts, at 8 ohms, both channels driven—or in the case of quadraphonic amps, with all channels driven.

For more information on any product, or on any product which is not listed because of our space limitations, the reader is invited to write to the manufacturer at the address listed below.

Directory of Manufacturers

Ace Audio Co.
25 Aberdeen Dr.
Huntington, NY 11743

Acoustic Research, Inc.
24 Thorndike St.
Cambridge, MA 02141

Acoustical Mfg. Co.
Huntingdon, England PE187DB

Advent Corp.
195 Albany St.
Cambridge, MA 02139

Akai America Ltd.
2139 E. DelAmo Blvd.
Compton, CA 90220

Allied-Radio Shack
2617 West 7th St.
Ft. Worth, TX 76107

Altec
1515 Manchester St.
Anaheim, CA 92803

Ampro Corp.
2220 Maplewood
Willow Grove, PA 19090

Astatic Corp.
Conneaut, OH 44030

Astrocom
Oneonta, NY 13820

Audio Devices, Inc.
100 Research Dr.
Glenbrook, CT 06906

Audio Dynamics Corp.
Pickett District Rd.
New Milford, CT 06776

Audio Research
2843 26th Ave. S.
Minneapolis, MN 55406

Audio-technica, U.S., Inc.
1655 W. Market St.
Fairlawn, OH 44313

Audioanalyst, Inc.
P.O. Box 262
Brookfield, CT 06804

Audionics
8600 N.E. Sandy Blvd.
Portland, OR 97220

Audioson-Kirksaeter
60 E. 42nd St.
New York, NY 10016

Avid Corp.
10 Tripps Lane
E. Providence, RI

B&O of America
2271 Devon Ave.
Elk Grove Village, IL 60007

BGW Systems
P.O. Box 3742
Beverly Hills, CA 90212

BSR (USA) Ltd.
Rte. 303
Blauvelt, NY 10913

Benjamin Electronics
40 Smith St.
Farmingdale, NY 11735

Bose Corp.
Framingham Ind. Park
Framingham, MA 01760

R.T. Bozak Mfg. Co.
Box 1166
Darien, CT 06821

British Industries Co. (BIC)
Westbury, NY 11591

CCA Electronics Corp.
716 Jersey Ave.
Gloucester, NJ 08030

Cerwin-Vega
6945 Tujunga Ave.
N. Hollywood, CA 91605

David Clark Co., Inc.
360 Franklin St.
Worcester, MA 01604

Creative Environments
85 Hoffman Ln. S.
Happauge, NY 11787

Crisman Speaker Co.
835 Walnut
Boulder, CO 80302

Crown International
Box 1000
Elkhart, IN 46517

Dathar Acoustics, Inc.
145 N. Franklin Tnpk.
Ramsey, NJ 07446

Dayton-Wright
P.O. Box 419
Thornhill, Ontario, Canada

Design Acoustics
P.O. Box 2722
Palos Verdes, CA 90274

Dokorder, Inc.
11264 Playa Ct.
Culver City, CA 90230

Dual (see United Audio)

DuKane Corp.
Commercial Sound Div.
St. Charles, IL 60174

Dynaco, Inc.
3060 Jefferson St.
Philadelphia, PA 19121

Eastman Sound Mfg. Co.
Harmony Rd.
Mickleton, NJ 08056

Elac (see Benjamin)

Electro Music
Bin 30, Arroyo Annex
Pasadena, CA 91109

Electro-Voice, Inc.
600 Cecil St.
Buchanan, MI 49107

Electronic Ind.
7516 42nd St. N.
Minneapolis, MN 55427

Electrostatic Sound Systems (ESS)
9613 Oates Dr.
Sacramento, CA 95827

Elite Electronics
195 Central Ave.
Farmingdale, NY 11735

Elpa Marketing
Thornes & Atlantic Aves.
New Hyde Park, NY 11040

Empire Scientific Corp.
1055 Stewart Ave.
Garden City, NY 11530

Epicure Products
1 Charles St.
Newburyport, MA 01950

Equasound
3330 S. Sepulveda Blvd.
Los Angeles, CA 90034

Ercona Corp.
2121 Bellmore Ave.
Bellmore, NY 11710

Fairfax Industries
900 Passaic Ave.
E. Newark, NJ 07029

Ferrograph (see Elpa)

Fisher Radio Corp.
11-40 45th Rd.
Long Island City, NY 11101

Frazier
1930 Valley View Lane
Dallas, Texas 75234

Garrard (see British Ind.)

Glenhurn Co.
787 Susquehanna Ave.
Franklin Lakes, NJ 07417

Gracom Industries
140-11A Cherry Ave.
Flushing, NY 11355

Grommes Div. Precision Electronics
9101 King St.
Franklin Park, IL 60131

HME (see Hill)

Hammond (see Microsound)

Harman-Kardon, Inc.
55 Ames Ct.
Plainview, NY 11803

Hartley Products
Box 68A
Hohokus, NY 10423

Heath Co.
Benton Harbor, MI 49022

Hegeman Laboratories
176 Linden Ave.
Glen Ridge, NJ 07028

Hervic Electronics
1508 Cotner Ave.
Los Angles, CA 90025

Hill Speaker Co.
P.O. Box 457
Lawrence, KS 66044

Hitachi Sales
48-50 34th St.
Long Island City, NY 11101

IMF Products
7616 City Line Ave.
Philadelphia, PA 19151

Impro Industries
120 Hartford Ave.
Mt. Vernon, NY 10553

Infinity Systems
9001 Fulbright Ave.
Chatsworth, Ca 91311

Innermedia Electronics
4503 E. Railroad Ave.
Sacramento, CA 95826

Integral Systems
463 Salem St.
N. Wilmington, MA 01887

JBL
3249 Casitas Ave.
Los Angles, CA 90039

JVC America
50-35 Queens Midtown Expy.
Maspeth, NY 11378

JansZen (see Electronic Ind.)

Jensen Sound Labs
4310 Trans World Rd.
Schiller Park, IL 60176

KLH Research & Development
30 Cross St.
Cambridge, MA 02139

Kenwood Electronics
15777 S. Broadway
Gardena, CA 90248

Klipsch & Associates
P.O. Box 280
Hope, AR 71801

Koss Electronics
P.O. Box 2320
Milwaukee, WI 53212

LWE, Div. Acoustron
7525 Wynlea
Houston, TX 77017

Lafayette
111 Jericho Tpke
Syosset, NY 11791

H.J. Leak (see Erocona)

Linear Devices
148 French St.
New Brunswick, NJ 08901

Linear Design Labs
114 Wilkins Ave.
Port Chester, NY 10573

Loudspeaker Design Corp.
2704 Garfield Ave.
Silver Spring, MD 20910

3M Company
2501 Hudson Rd.
St. Paul, MN 55119

Magnavox Corp.
345 Park Ave.
New York, NY 10022

Magnum Opus
220 W. 19th St.
New York, NY 10011

Marantz Co.
8150 Vineland
Sun Valley, CA 91352

Martin (see Eastman)

Maximus Sound
809 Stewart Ave.
Garden City, NY 11530

Microsound Co.
Box 4591
Colorado Springs, CO 80909

Miracord (see Benjamin)

Mura Corp.
50 S. Service Rd.
Jericho, NY 11753

Nagra Magnetic Recorders
19 W 44th St.
New York, NY 10036

Nakamichi Research
220 Westbury Ave.
Carle Place, NY 11514

North American Philips
100 E. 42nd St.
New York, NY 10010

Ohm Acoustics
133 Emerson Place
Brooklyn, NY 11205

Olson Electronics
260 Forge St.
Akron, OH 44308

Onkyo
25-19 43rd Ave.
Long Island City, NY 11101

PE (see Impro)

PML (see Ercona or Hervic)

P&M Electronics
519 S. Austin
Seattle, WA 98108

Pacific Electronics
6601 Bay St.
Emeryville, CA 94608

Panasonic (see Technics)

Paoli Hi-Fi
P.O. Box 876
Paoli, PA 19301

Phase Linear
405 Howell Way
Edmond, WA 98177

Pickering & Co.
Sunnyside Blvd.
Plainview, NY 11803

Pilot
66 Fieldpoint Rd.
Greenwich, CT 06830

U.S. Pioneer Electronics
178 Commerce St.
Carlstadt, NJ 07072

QRK (see Ercona)

Quadraflex (see Pacific)

Quintessence
1626 N. "C" St.
Sacramento, CA 95814

RTR Industries
8116 Deering Ave.
Canoga Park, CA 91304

Rabco
55 Ames Ct.
Plainview, NY 11803

Radio Shack (see Allied-Radio Shack)

Rectilinear Research
107 Bruckner Blvd.
Bronx, NY 10454

Revox Corp.
155 Michael Dr.
Syosset, NY 11791

Rogersound Labs
6319 Van Nuys Blvd.
Van Nuys, CA 91401

Rotel of America
2642 Central Park Ave.
Yonkers, NY 10710

Sansui Electronics
55-11 Queens Blvd.
Woodside, NY 11377

Sanyo Electric
1200 W. Walnut St.
Compton, CA 90220

Schober Organ Corp.
43 W. 61st St.
New York, NY 10023

Scientific Audio Electronics (SAE)
P.O. Box 60271, Terminal Annex
Los Angeles, CA 90060

Scintrex Audio
Amherst Ind. Park
Tonawanda, NY 14150

Sennheiser Electronics
10 W. 37th St.
New York, NY 10018

Sherwood Electronic Labs
4300 N. California
Chicago, IL 60618

Shure Brothers
222 Hartrey Ave.
Evanston, IL 60204

Sony Corp. of America
47-47 Van Dam St.
Long Island City, NY 11101

Soundcraftsmen
1310 E. Wakeham
Santa Ana, CA 92705

Southwest Technical Products
219 W. Rhapsody
San Antonio, TX 78216

Stanton Magnetics
Terminal Dr.
Plainview, NY 11803

Superex Electronics
151 Ludlow St.
Yonkers, NY 10705

Superscope, Inc.
8150 Vineland Ave.
Sun Valley, CA 91352

Sylvania
700 Ellicott St.
Batavia, NY 14020

Tandberg of America
8 Third Ave., Box 171
Pelham, NY 10803

Tannoy (America) Inc.
1756 Ocean Ave.
Bohemia, NY 11716

Tascam Corp.
5440 McConnell Ave.
Los Angeles, CA 90066

TEAC Corp. of America
7733 Telegraph Rd.
Montebello, CA 90640

Technics by Panasonic
200 Park Ave.
New York, NY 10017

Telex Communications
9600 Aldrich Ave. S.
Minneapolis, MN 55420

Thorens (see Elpa)

Tomlinson Research
1890 Capitol Circle, SW
Tallahassee, FL 32301

Toshiba America
4106 Delong St.
Flushing, NY 11355

Toyo
1842-B W. 165th St.
Gardena, CA 90247

Transduction Ltd.
P.O. Box 608
Bristol, PA 19007

Trusonics
1100 E. Franklin
Huntington, IN 46750

The Turner Co.
909 17th SE
Cedar Rapids, IA 52402

Unicord Inc.
75 Frost St.
Westbury, NY 11590

United Audio
120 S. Columbus Ave.
Mt. Vernon, NY 10553

Utah Electronics
1124 E. Franklin St.
Huntington, IN 46750

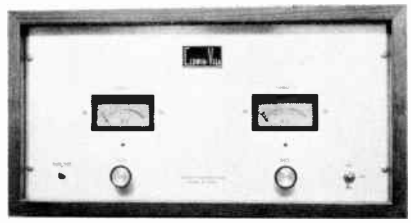
Venturi (see BIC)

V-M Corp.
Box 1247
Benton Harbor, MI 49022

Yamaha
6600 Orangethorpe Ave.
Buena Park, CA 90620

Amplifiers–Basic & Integrated

Bose 1801



Cerwin-Vega A-3000S

Crown DC-300A

Notes: (1) All models solid-state except when model number is preceded by (T).
(2) Basic power amplifiers have model number preceded by (B).
(3) "K" indicated kit price; "W" wired.

MANUFACTURER	MODEL	RMS power/chan., W, 8 ohms	THD at rated power, %	THD at 1 watt, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz - kHz	Freq. resp. at 1 watt, Hz ±? dB	Rated output S/N, phono, dB	Phono sensitivity, mV	Phono overload, mV	Tape head input, mV	High level input, V	Output Z, ohms	Damping factor	Dimensions, w x d x h, in.	Weight, lbs.	Price	SPECIAL FEATURES
ACOUSTICAL MFG.	(B) Quad 303	45	.03	.03			20-30k	30-35k −1	100					8		4¾ x 12¾ x 6¼	18	237.00	
ALTEC	770A	90	0.3				15-25k	20-20k ± 0.5					0.5	8	25	6½ x 9⅞ x 9	16	284.00	Built-in elect. x-over switchable to 800 or 500Hz; 2 basic pwr. amps on 1 chassis.
AUDIO RESEARCH	(BT) D51	50	0.1	0.01	0.5	0.01	15-30k	5-20k ± 1	90					4,8, 16	20	19 x 13 x 7		695.00	
	(BT) D75	75	0.1	0.01	0.5	0.01	15-30k	5 20k ± 1	90					4.8, 16	20	19 x 13 x 7		995.00	
	(BT) D400	400	0.1	0.01	0.5	0.01	15-30k	5-20k ± 1	90					4,8, 16	20	19 x 13 x 10½		2950.00	3 chassis; built-in fans; meters.
BGW SYSTEMS	(B) 1000	250	0.1	0.1	0.1	0.1	5-20k	2-80k +0 −3	100				2	4	1000	19 x 17 x 7	70	1200.00	SCR crow bar; no fuses; adj. power limit.; forced air cool.; LEDs
	(B) 4X250 4-chan.	250	0.1	0.1	0.1	0.1	5-20k	2-80k +0 −3	100				2	4	1000	19 x 17 x 7	70	1450.00	4-channel; sim. to above; anti-trans. ON-OFF; SCR crow bar.
	(B) 500R	200	0.2	0.2	0.1	0 1	5-20k	2-65k +0 −3	100				2	4	1000	19 x 11 x 5¼	42	685.00	Mod. const,; SCR crow bar.
	(B) 250	100	0.2	0.2	0.1	0.1	5-20k	2-65k +0 −3	100				2	2	1000	19 x 8 x 5¼	30	429.00	Will drive 2 ohm loads; for use with sm. sys. & multiple speakers.
	(B) 1500	1500*	0.1	0.1	0.1	0.1	5-10k	5-20k +0 −3	100				2	1	2000	19 x 17 x 7	75	1350.00	*Mono–1500 watts 1 ohm; high power bass reprod.
BOSE	(B) 1801	250	0.2	0.2	0.2	0.2	5-50k	20-20k ± 0.4					1.5	4	200	18 x 18½ x 7¼	82	986.00	Mtrs.; LED display; 2 inputs; 2 outputs; 400 W. into 4 ohms.
	(B) 1800	250	0.2	0.2	0.2	0.2	5-50k	20-20k ± 0.4					1.5	4	200	19 x 15 x 8¼	80	1000.00	LED display; rack mountable: field serviceable; 400 W. into 4 ohms.
CERWIN-VEGA	A-3000	400	0.08	0.08	0.08	0.1	20-20k	5-70k ± 1	110	1.8V				2	500	19 x 9 x 9	55	895.00	
	A-2800	700	0.08	0.08	0.08	0.1	20 20k	5-70k ± 1	110	1.8V				2	500	19 x 6½ x 9	50	850.00	Short ckt. output protect.; avail. w/elect. x-over; mono.
CRISMAN	Monolith Mk XI	100	0.1	0.1	0.2	0.2	10-100Hz	10-100Hz ± 5						0.1	150	42 x 20 x 26	200	1149.00	Incl. 30" woofer in air susp. cab.
CROWN	(B) D-60	30	0.05	0.05	0.05	0.05	5-30k*	20-20k 0.1	106				0.775	4,8, 16	200	17 x 18¾ x 1¾	10	249.00	* ± 1 dB; wal. cab., $29.00; frnt. pnl. phone jack; mono conv. to 90 W. 25 volt line out (bal.).
	(B) D-150	75	0.05	0.05	0.05	0.05	10-20k*	4-100k ± 1	110				1.19	4,8, 16	200	16½ x 8 x 5	22	399.00	* ± 1 dB; wal. cab., $33.00; frnt. pnl, $30.00; mono conv. to 250 W. 50 volt line out (bal.).
	(B) DC-300A	150	0.05	0.05	0.05	0.05	DC-20k*	DC-100k ± 1	110				1 75	1-16	200	19 x 9¾ x 7	45	695.00	* ± 1 dB; wal. cab., $37.00; mono conv. to 600 W. 70 volt line out (bal.).
DAYTON WRIGHT	(B) DWIK	500	0.06	0.03	0.09	0.03	10-21k	10-25k ± 5							500	19 x 14 x 7	92	1460.00	All output dev. ind. fused; each chan. (incl. output) plug-in cond.; fan cooled, avail. 11/1/73.
DUKANE CORP.	1A901B	50	1.5	0.5				20-20k ± 1								19 x 6½ x 5¼	25		Circuit protecting devices.
	1A911B	100	1.5	0.5				20-20k ± 1								19 x 7 x 8¾	36		Same as above.
	1A921B	200	1.5	0.5				20-20k ± 1								19 x 8 x 12¼	55		Same as above.

Amplifiers-Basic & Integrated

Dynaco ST-400

Heath AA-2010

JVC VN-900

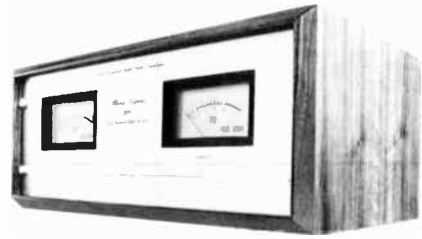
Notes: (1) All models solid-state except when model number is preceded by (T).
(2) Basic power amplifiers have model number preceded by (B).
(3) "K" indicated kit price; "W" wired.

MANUFACTURER	MODEL	RMS power/chan., W, 8 ohms	THD at rated power, %	THD at 1 watt, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz - kHz	Freq. resp. at 1 watt, Hz - ? dB	Rated output S/N, phono, dB	Phono sensitivity, mV	Phono overload, mV	Tape head input, mV	High level input, V	Output Z, ohms	Damping factor	Dimensions, w x d x h, in.	Weight, lbs.	Price	SPECIAL FEATURES
DYNACO	Stereo 400 (B)	200	0.18	0.05	0.1	0.05	5-35k	8-50k ±0.1	95				1.6	8	80	17 x 14 x 7	54	449.00K 599.00W	Spkr. prot.; relay d.c. load prot.; 1000 sq. in. heat sink; opt. output mtrs., filts., lev. contls.
	SCA-80Q 4-chan.	40	0.5	0.1	0.5	0.1	8-50k	15-50k ±½	60	3.0	80		0.13	8	40	13½ x 10 x 4	16	169.95K 249.95W	Built-in matrix cirt. for 4-D sound w/4 spkrs
	Stereo 120 (B)	60	0.5	0.1	0.5	0.1	5-50k	5-100k ±½	100				1.5	8	40	13 x 10½ x 4	20	159.95K 199.95W	Regulated pwr. supply.
	Stereo 80 (B)	40	0.5	0.1	0.5	0.1	8-50k	10-50k ±½	90				1.3	8	40	14 x 8 x 4	13	119.95K 159.95W	
ESS	500	250	0.1	0.1	0.1	0.1	1-100k	3-150k ±1							1000	16⅝ x 15 x 6		625.00	Active constant current source ctry. no turn-on surges; 1.7 μS rise time.
EPICURE		125	0.1	0.05	0.1	0.05	5-35k	3-100k ±3	100					4-8	100	13 x 19 x 8			200 W. into 4 ohms spec. protect circ.
HARMAN/ KARDON	Citation 12 Deluxe	60	0.2	0.2	0.15	0.15	5-35k	½-100k ±1	100							12¼ x 12⅝ x 5½	32	340.00	W. face plate & wood encl; Citation 12 less faceplate & encl., $295.00W, $225.00K.
HEATH	AA-14	10*	1.0*	0.5	1.0		7-90k	12-60k ±1	60	4.5			0.3		50	12 x 10¼ x 3	8½	59.95K	Stereo 'phone jack; spkr. swit.; *full power 20-20k at rated THD (all channels driven).
	AA-1214	15*	0.5*	0.25	0.5	0.2	5-30k	7-100k ±1	65	2.0	75		0.19		50	12¾ x 12 x 3⅞	10	89.95K	*Pwr rating as above; tape mon. input; stereo 'phone jack; spkr swit.
	AA-29	35*	0.25*	0.1	0.2	0.1	5-30k	7-60k ±1	65	2.2	155		0.18		50	16¾ x 14½ x 5⅛	22½	159.95K	*Pwr. rating as abuve; inud. constr.; 6 inputs incl. tape mon.; 2 spkr. swit.
	AA-15	50*	0.5*	0.2	0.5	0.2	6-30k	8-40k ±1	60	2.2	155		0.2		45	16⅞ x 12½ x 4¾	21½	189.95K	*Pwr. rating as above; 5 inputs incl. tape mon.; tone flat; loudness; 2 spkr. swits.
	AA-2010 4-chan.	35*	0.25*	0.1	0.2	0.1	5-45k	7-50k ±1	65	2.2	155		0.18		100	18½ x 13¾ x 6½	28	359.95K	*Pwr. rating as above; 4-ch. amp (incorp. 2 compl. stereo amps); 4 meters w/3-pos. atten swit.; handles all matrix material.
HITACHI	IA-600	32	0.5				20-30k	10-100k +0, −1	95	2.5			0.22	8	50	16 x 5 x 12	19	219.95	
	IA-1000	55	0.5				20-35k	10-100k +0, −1	100	0.25 2.0			0.14	8	50	18 x 5 x 13	28	329.95	
INTEGRAL SYSTEMS	(B) 200	100	0.2	0.2	0.2	0.2	20-25k	10-200k ±1							100	19 x 12⅞ x 5¼		300.00	Model (B) 200U, utility, $270.00
JVC	4VN770 4 chan.	12.5	0.5	0.2	0.8	0.3	10-30k	18-40k ±1	65	2.5	100		0.1	4-16	50	16½ x 12 x 5½	21½	279.95	4 chan. integ. amp; 4 VU mtrs.
	4VN990 4 chan.	35	0.5	0.1	0.8	0.2	10-30k	10-50k ±1	65	2.5	100		0.15	4-16	50	16⅝ x 15⅜ x 5⅜	35¼	499.95	4 chan. integ. amp; 4 VU mtrs.; dual SEA contls.
	VN700	35	.25	.05	0.4	0.1	25-20k	20-50k ±.5	65	2.5	170		0.2	4-16	50	16⅝ x 12½ x 5⅜	22	299.95	SEA tone contls
	VN900	50	.25	.05	0.4	0.1	20-20k	20-50k ±0.5	65	2.5	250		0.2	4-16	50	16⅝ x 12½ x 5⅜	28	369.95	SEA tone contls; pink noise tester
	(B) VB-10	60	.07	.05	0.1	.05	10-70k	10-100k ±0.2					1.0	4-16	80	19 x 13½ x 6	36	599.95	2 VU mtrs.
	(B) VB-100	50	.07	.05	.15	0.1	20-30k	18-45k ±0.5					0.8	4-16	0.5-50	8½ x 12½ x 5½	16½	259.95	2 VU mtrs.; var damp.

Amplifiers-Basic & Integrated

Kenwood KA-8004

Marantz 250



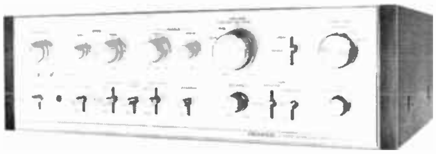
Phase Linear 400

Notes: (1) All models solid-state except when model number is preceded by (T).
(2) Basic power amplifiers have model number preceded by (B).
(3) "K" indicated kit price; "W" wired.

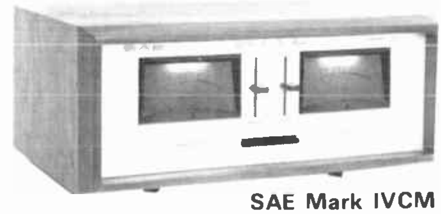
MANUFACTURER	MODEL	RMS power/chan., W, 8 ohms	THD at rated power, %	THD at 1 watt, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz - kHz	Freq. resp. at 1 watt, Hz ± dB	Rated output S/N, phono, dB	Phono sensitivity, mV	Phono overload, mV	Tape head input, mV	High level input, V	Output Z, ohms	Damping factor	Dimensions, w x d x h, in.	Weight, lbs.	Price	SPECIAL FEATURES
KENWOOD	KA-8004	60	0.4		0.4		10-50k	20-50k −2	65	2.5	200				30	17⅛ x 11¾ x 6	28.4	389.95	Dir. cplg.; dual protect. crt.; bass & treb. contls. w/sel. x-over pts.; 2 sys. tape; A-B-C spkr.
	KA-6004	40	0.5		0.3		10-50k	20-40k +0, −1	68	2.5					32	17⅛ x 11¾ x 6	25.4	299.95	Dir. cplg.; dual protect.; PNP can-type trans.; 2 sys. tape; 40 & 80 lo filters; A-B spkr. sys.
	KA-4004	36	0.5		0.5		10-50k	20-40k +0, −1.5	65	2.5					32	17⅛ x 11¾ x 6	20½	189.95	Dir. cplg.; dual protect. crt.; PNP trans.; 2 sys. tape; inputs for 2 phono, 2 aux, tuner, mic A-B spkr.
	KA-2002A	13	0.8		0.8		20-30k	20-30k ± 2	60	2.0					50	13½ x 9½ x 4⅝	11.9	119.95	Bass & treb. contls.; inputs for 2 phono, tuner, aux, tape.
LAFAYETTE	LA-74 4-chan.	30	1	0.07			18-25k	20-20k ± 1.5	65	3.7			0.55			15⅝ x 11¾ x 4⅛	18.7	279.95	Full logic SQ crty. w/Variblend.
	LA-222 4-chan.	7	1	0.15				20-20k ± 1.5	55	2.5	55		0.27	4, 8, 16		13½ x 8¾ x 4⅜	14	129.95	SQ crty.; main/rem. spkr. swit.
	LA-150	33	1	0.05			13-35k	22-20k ± 1	56	2.2, 7	40, 120		0.25	4, 8, 16		13 x 9¼ x 3⅞	19	169.95	Main/rem. spkr. swit.; lev. contls
	LA-975 4-chan.	25	1	0.07			20-35k	20-20k ± 1	60	4.5	60		0.25	4, 8, 16		13 x 10½ x 4¼	20	149.95	SQ crty.
H.J. LEAK (ERCONA)	Stereo 70	35	0.1	0.1	0.3			20-30k	66	2.0		2.0		8	40	13 x 4¼ x 8¾	12	299.00	7 pushbutton controls; Mono "L" & "R" inputs; headphone output; remote & main spkrs.
MARANTZ	500	250	0.05	0.05	0.05	0.05	3-60k	20-20k ± 0.1						4-16	400	17⅝ x 16 x 7	81.3	1200.00	Output mtrs.; pwr. limit. swit. for 50W., 150W., 250W.
	250	125	0.1	0.1	0.1	0.1	7-60k	20-20k ± 0.1						4-16	100	15⅝ x 9½ x 6⅛	31	495.00	Output mtrs.; mtr. rng. swit. for −20dB; 240 same but w/o output mtrs. or mtr rng. swit., $395.00.
	1200	100	0.15	0.15	0.15	0.15	10-40k	20-20k ± 0.25	100*	1.35	100		0.134	4-16	100	15⅝ x 14 x 5¾	34	595.00	*Phono dyn. rng.: ratio of overload to equiv. input noise.
	1120	60	0.2	0.2	0.2	0.2	10-40k		96*	1.1	100		0.11	4-16	40	15⅝ x 14 x 5¾	32½	395.00	*Same as above.
	1060	30	0.5	0.5	0.5	0.5	15-50k	20-20k ± 1	96*	1.8	100		0.15	4-16	45	14¼ x 12 x 4¾	26	229.95	*Same as above.
	1030	15	0.5	0.5	0.5	0.5	15-40k	20-20k ± 1	93*	2.1	100		0.18	4-16	40	14¼ x 12 x 4¾	25	169.95	*Same as above.
	4140 4-chan.	25**	0.3	0.3	0.3	0.3	7-70k	20-20k ± 0.5	96*	1.9	100		0.18	4-16	40	15⅝ x 14 x 5¾	40.7	549.95	**4/ch., 70W./chan. in 2-chan. mode; 2-ch./4-ch. bridging; Vari-Matrix synth. w/dim. contl.; rem. contl. capability; *same as above. 4100 same but 60W./chan. in 2-ch. mode.
	4070 4-chan.	15**	0.5	0.5	0.5	0.5	10-60k	20-20k ± 1	93*	1.9	100		0.15	4-16	40	14¼ x 12 x 4¾	27	299.95	**4-chan., 35W/chan. in 2-chan. mode. Balance same as above.
OLSON	AM 395	12	0.75	0.4	1.5	0.52	20-28k	20-20k ± 1.8	58	2.0	40	2.8	0.2	4-16	20	11½ x 7¼ x 4¼	12	69.99	
	AM 372	8	1.75	1.5	1.5	1.0	22-25k	20-20k ± 3	55	2.0	45	3.0	0.25	4-16	18	12¼ x 7½ x 3¼	7	39.98	
ONKYO	7055	26	0.1	0.03	0.05		10-100k	10-70k to −1	75	1.2/2.4/4.8	230			8/4	80	16⅝ x 4⅛ x 5⅜	25	219.95	3 pos. phono gain switch; 250 prot. circuitry.
	7022	52	0.1	0.03	0.05		10-100k	10-70k +0 −1	75	1.2/2.4/4.8	300			8/4	80	16⅝ x 14¾ x 5⅜	29	329.95	-300 mV phono overload capacity; sep. swit. between pre and main amp.; choice of turnover freq.
P & M	(B) S-100	120	0.3				2-90k	3-55k ± 1							50	15½ x 11 x 5	29	369.50	W. 2 VU mtrs. Mono-100, $199.50.
	(B) S-50	50	0.5				10-45k	15-50k ± 1							40	15½ x 10 x 3	14	199.95	Mono-50, $124.50.
	(B) S-25	25	0.5				15-45k	20-150k ± 1							40	7⅛ x 7 x 3	10	129.50	Mono-25, $79.50.
PHASE LINEAR	400	200	0.25	0.25	0.25	0.25	5-20k	5-250k ± 0	100					4-16	1000	19 x 10 x 7	35	499.00	
	700	350	0.25	0.25	0.25	0.25	0-20k	0-250k ± 0	100					4-16	1000	19 x 10 x 7½	45	799.00	

Amplifiers–Basic & Integrated

Pioneer SA-9100



Rotel RA-1210



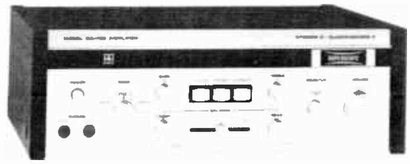
SAE Mark IVCM

Notes: (1) All models solid-state except when model number is preceded by (T).
(2) Basic power amplifiers have model number preceded by (B).
(3) "K" indicated kit price; "W" wired.

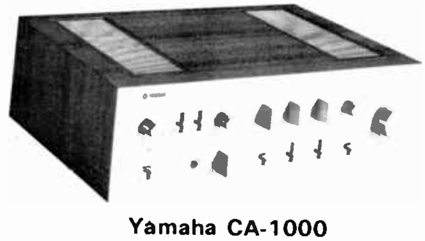
MANUFACTURER	MODEL	RMS power/chan., W, 8 ohms	THD at rated power, %	THD at 1 watt, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz - kHz	Freq. resp. at 1 watt, Hz ±1 dB	Rated output S/N, phono, dB	Phono sensitivity, mV	Phono overload, mV	Tape head input, mV	High level input, V	Output Z, ohms	Damping factor	Dimensions, w x d x h, in.	Weight, lbs.	Price	SPECIAL FEATURES
PILOT	210	25	0.5		0.5		20-25k		65	2.5 4.5				4,8, 16	35	15 x 11 x 5⅛		159.90	
	310	30	0.5		0.5		10-40k		70	2.5 4.5				4,8, 16	35	18½ x 17½ x 6½		349.90	4 Mtrs; Pilotone oscillator; mic mixing.
PIONEER	SA-5200	13	0.8	0.2	0.8	0.4	10-40k			2.5	100			8	30	16 x 12 x 5	16	139.95	2 aux.; 2 spkr. sys.; click-stop tone contls.
	SA-7100	22	0.5	.05	0.5	0.01	5-70k	7-80k +0, −1		2.5	190			8	40	17 x 13 x 5	22	249.95	2 phono; 2 aux.; 2 tape mon. dup. 1-2, 2-1; tone defeat swit ; freq. turnover swits.; −20 dB cut muting.
	SA-8100	44	0.3	0.05	0.3	0.05	5-40k	7-80k ± 1		2.5	250			8	60	17 x 13 x 5	26	349.95	Twin tone contls, main & sub.; vol. lvl. set contl.; tone defeat swit.; 2 tape mon., 1-2, 2-1; 2 aux., 2 phono: −20 dB cut muting.
	SA-9100	65	0.1	0.04	0.1	0.04	5-40k	7-80k ± 1		2.5	250			8	70	17 x 13 x 5	30	449.95	Twin tone contls, main & sub.; vol. level set contls.; tone defeat swit. 2 tape mon., 1-2, 2-1; 2 aux., 2 (phono 2 imp. sel.); −20 dB muting cut; aux 2 phono 2 & spkr. B level contls.
QUINTESSENCE	(B) Power Amp II	120	0.05	0.05	0.05	0.05	10-45k	10-500k ± 0.25					0.5	8	1000	17 x 10 x 6	55	750.00	Lev. sens.; bal., compl. sym. drivers & output; op-amp 1st stg.; fan cooled.
	(B) Power Amp IID	120	0.05	0.05	0.05	0.05	10-45k	10 500k ± 0.25					0.5	8	1000	17 x 10 x 6	60	1000.00	Same as above w/liq. crys. dig. watt mtr., reading peak or RMS lev.
RADFORD (AUDIONICS)	(B) HD-60	60	0.005	0.005	0.02	0.02	10-70k						1.0	4-16	50	16 x 4½ x 8¾	22	429.95	
	HD-2260	60	0.01	0.01	0.02	0.02	10-70k		80	1.0	750		0.08	4-16	50	16 x 4½ x 8¾	30	na	
RADIO SHACK	QA-621 4-chan.	2½					30-20k	20-25k ± 3		150			140	4-16		12 x 8 x 4		79.95	SQ
	QA-681	25					30-20k	20-25k ± 3						4-16		14⅛ x 11¾ x 4		199.95	SQ
	SA-500	30	1				30-20k	20-20k ± 1						4-16		14¾ x 11 x 3⅝		119.95	
REVOX	A78	40	0.1	0.1	0.3		10-40k	20-20k ± 1	80	2.0			0.25	4-16	20	16¾ x 7⅝ x 6¼	18	449.00	Step tone contls ; sep adj inputs; 2 phone outputs; ind., low, hi & pres.
ROTEL	RA-211	10	0.2		0.5	1.1	20-35k	20-75k	63	2.7	50	8.0		8	35	11¾ x 7⅛ x 4	10	109.95	Spkr. sys.: 1-2, 1+2; tape mon.; wood cab.
	RA-311 4-chan.	18	0.1		0.3		15-50k	15-90k	65	2.7	50			8	35	12¾ x 8¾ x 4	12	159.95	Spkr. sys.: 1-2, 1+2; 4-ch. syn.; hi fil.; tape mon. 1+2; tape dub. 1+2, wood cab.
	RA-611	30	0.1		0.2		5-50k	10-100k ± 0.5	65	2.5	100			8	35	16½ x 12 x 5½	14	239.95	Dual slide bass & treble; tape mon. 1-2; tone defeat; loudness; phono 1+2; Aux. 1-2; low & hi fil.; wood cab.
	RA-810	40	0.2		0.1		5-50k	4-75k +0, −3	65	2.0	125			8	38	16½ x 12 x 5½	22	299.95	Tape dub. 1-2, 2-1; adj. phono imp./sens. on phone 2; hi-lo fil.; 20dB muting.
	RA-1210	60	0.05 to 0.09	0.02	0.1		3-100k +0, −3	70	1.0	130				8	38	16½ x 12 x 5½	30	349.95	2 power transf.-adj. imp. & sens. on phono 2; tape dub. 1-2, 2-1; 3 stage diff. preamp; bass & treble roll-off.
SAE	(B) MK III CM	200	0.05	0.05	0.025	0.025	8-50k	1-100k ± 1	100						150+	17 x 5¾ x 14½	49	950.00	Series D output; dir. rdg. pwr. mtrs. in rms; Mk IIIC same but w/o pwr. mtrs., $750.00.
	(B) Mk IV CM	100	0.05	0.05	0.025	0.025	8-50k	5-100k ± 1	100						150+	17 x 15 x 5¾	33	550.00	VU mtrs.; Mk IVC same but w/o VU mtrs., $500.00
	(B) Mk 31B	50	0.05	0.05	0.025	0.025	8-50k	5-100k ± 1	100						150+	15 x 8 x 4¾	18	275.00	Series D output.
	(B) Mk 31 Mono	60	0.05	0.05	0.05	0.05	8-50k	5-100k ± 1	100						150+	15 x 8 x 4¾	16	150.00	

Amplifiers–Basic & Integrated

Sherwood S-9400



Superscope QA-450



Yamaha CA-1000

Notes: (1) All models solid-state except when model number is preceded by (T).
(2) Basic power amplifiers have model number preceded by (B).
(3) "K" indicated kit price; "W" wired.

MANUFACTURER	MODEL	RMS power/chan., W, 8 ohms	THD at rated power, %	THD at 1 watt, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz - kHz	Freq. resp. at 1 watt, Hz ± ? dB	Rated output S/N, phono, dB	Phono sensitivity, mV	Phono overload, mV	Tape head input, mV	High level input, V	Output Z, ohms	Damping factor	Dimensions, w x d x h, in.	Weight, lbs	Price	SPECIAL FEATURES
SANSUI	AU9500	85	0.1	0.1			5-40k	15-50k +0 -0.5		2.5		100	0.8	4, 8	50	19⅝ x 13⅝ x 5½	51¼	519.95	
	AU7500	43	0.1	0.1			5-40k	10-50k ±1		2.5		100	0.8	4, 8	40	17⅜ x 12⅝ x 5½	28	299.95	6500 sim. but 32W/chan. & freq. resp. 10-40k, $249.95.
	(B) BA-60	20	0.3	0.3			20-40k	20-60k ±1					0.7	4-16	46	6 x 10⅜ x 4⅜	9-1/6	99.95	
	QA7000 4-chan.	20	0.1				10-30k	20-40k ±0.5		2.5	270	100	0.8	4-16	100	17⅜ x 12¾ x 5½	30.9	549.95	Decoder; signal avail. at Tape 2; dir. dubbing.
SCHOBER	BTR-3M	70	0.09	0.08	0.07	0.05	5-40k	5-57k ±0.5					0.15 to 1.0	4-16	28	5½ x 11⅜ x 8	20	na	Mono, conv. to 2-chan.; fully protected.
	BTR-3D	70	0.09	0.08	0.07	0.05	5-40k	5-57k ±0.5					0.15 to 1.0	4-16	28	5½ x 11⅜ x 8	21	na	Stereo; fully protected.
SHERWOOD	S-9400 4-chan.	50	0.8	0.15	0.6	0.3	5-45k	20-20k ±0.5	60	1.8	80	2.1	0.2	8	40	17½ x 14 x 5½	29	259.95	2 phono; 2 aux; mics; Dynaquad; spkr. overload prot.
SINCLAIR (HERVIC)	2000	8	0.06		0.1		25-25k	25-35k ±3	65	3.0 & 30.0			0.125	8	55	12 x 6 x 2	3	129.50	Push-button contls.; and blk. alum. case; 3000 same w/17 W./chan.; THD at rtd. pwr., 0.04.
SONY	TA 1055	23	0.5	0.1	0.5	0.2	10-40k	10-60k ±3	70	2.0	70		0.25	8	22	16⅛ x 11⅛ x 4¾	13½	189.50	20 + 20 W. at 40-20k; wood cab.; 2 tape mon.; dir. spkr. coupling.
	TA 1150	35	0.2	0.1	0.2	0.1	8-35k	15-80k ±2	70	2.0	70		0.14	8	100	15¾ x 12½ x 5¾	18¼	269.50	30 + 30 W. at 20-20k; 2 tape mon.; dir. spkr. coupling.
	TA 1130	65	0.1	0.05	0.1	0.05	7-30k	10-100k ±1	70	1.2	70		0.13	8	100	15¾ x 12⅞ x 5⅞	28¾	429.50	50 + 50 W. at 20-20k; 2 dB-step calib. tone contls.; dir. spkr. coupling.
	(B) TA 3130F	70	0.1	0.05	0.1	0.05	7-30k	10-200k ±1					1.0	8	200	7⅞ x 12¾ x 5⅞	17¼	249.50	50 + 50 W. at 20-20k; dir. spkr. coupling.
	(B) TA 3200F	110	0.1	0.03	0.1	0.03	5-35k	5-200k ±1					1.4	8	200	15¾ x 12¾ x 5⅞	28	369.50	100 + 100 W. at 20-20k; ½, ¼ pwr. limit. swit.; dir. spkr. coupling.
SOUTHWEST TECHNICAL	(B) 540	15	1.0	0.2	1.5	0.5	10-50k	5-100k					1.0	8	100	9 x 8 x 2½	6	39.95K	Integ. crt. driver.
	(B) 185	30	0.1	0.05	0.5	0.1	10-80k	5-150k					1.25	4.8	100	10 x 9 x 3	8	55.00K	Current src. driver sys.; compl. outputs.
	(B) 175	80	0.1	0.02	0.3	0.1	10-100k	5-200k					1.5	4.8	100	13 x 6 x 5	14	60.00K	Mono; current src. driver; compl. output stg. w/volt-amp limit sys.
	(B) 207	60	0.02	0.015	0.01	0.005	10-100k	5-250k					1.5	4.8	100	13 x 4 x 5	13	75.00K	Mono; all compl. push-pull x-cpld. crt.; triple output stg.; volt-amp limiting.
SUPERSCOPE	A-235	5	1.0				30-50k			3.0						14⅓ x 7⅜ x 4½	6.6	79.95	Mag/cer phono swit.
	A-245	10	1.0				13-23k			2.5						14⅓ x 7⅜ x 4½	8	99.95	Tape mon.; mag/cer phono swit.; loudness swit.
	A-260 4-chan.	15	0.05				20-50k			2.0						14½ x 11¾ x 5⅛		169.95	Quadraphase (spkr. matrix sys.); loudness contl.; tape mon.; detented sliding tone contls.; mic jack for p.a.
	QA-450 4-chan.	10*	1.0				30-30k			3.5						14½ x 12⅜ x 5¼		279.95	*4-chan., 25W/chan. in 2-chan. mode; discrete, SQ; ambience: 2ch/4 ch. bridging; swit. preamp/pwr. amp modes; loudness; tape mon.; hi freq. filter swit.; 4-ch. vol. contl.

MANUFACTURER	MODEL	RMS power/chan., W, 8 ohms	THD at rated power, %	THD at 1 watt, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz - kHz	Freq. resp. at 1 watt, Hz ± dB	Rated output S/N, phono, dB	Phono sensitivity, mV	Phono overload, mV	Tape head input, mV	High level input, V	Output Z, ohms	Damping factor	Dimensions, w x d x h, in.	Weight, lbs	Price	SPECIAL FEATURES
TOMLINSON	1002	100	0.1	0.05	0.1	0.05	10-25k	10-100k ±1							500	17 x 11¾ x 7	42	559.95	Level contl; short-proof protect. circuit.
	3501	350	0.1	0.05	0.1	0.05	10-25k	10-100k ±1							500	17 x 11¾ x 7	42	599.95	Similar to above—mono.
V-M	1538	4	5.0	0.8				70-10k	55				0.1	8					2 4-chan. discrete inputs; 2 E-V decoded stereo inputs.
YAMAHA	CA 600	30	0.1	0.04	0.1	0.05	5-70k	10-50k +0.5 −1.0	80*	3.0	310		0.12	4-8	70	17 x 6 x 13	25	299.95	*IHF A ntwrk.; OCL pure compl., cont. loud. contl.; trans. & spkrs. prot.; 2 tape dub. ckt.
	CA 800	45	0.1	0.04	0.1	0.05	5-70k	10-50k +0.5 −1.0	80*	3.0	310		0.12	4-8	70	17 x 6 x 13	31	469.95	*As above; sel. tone contl.; turnover freq.; 2-step hi & lo fil.
	CA 1000	70	0.1	0.04	0.1	0.05	5-50k	10-50k +0.5 −1.0	80*	3.0	310		0.12	4-8	70	17 x 6 x 13	34	569.95	*As above; FET phono input; spec. mic amp; 3 spkrs output.

Amplifier Q's and A's-- Mainly For Beginners

Q. *Is it still necessary for amplifiers to be divided into a preamp and a power unit? Why should I be bothered with all those connecting cables?*

A. Years ago, in the tube era, were several advantages in having separate units. For some cabinet installations, the main amplifier with its massive power and output transformers could be placed at the bottom, leaving the control unit to be mounted in the most convenient position. Secondly, the radiated hum field from the power transformer was more difficult to control in an integrated unit. Thirdly, the sheer size of a high-power integrated unit was a disadvantage in itself. But with solid-state techniques these advantages have tended to disappear. Power transformers can be much smaller and there are no output transformers to contend with. And so, integrated amplifiers with powers up to 100 watts per channel are every bit as good as separates. But for very high powers—of the order to two or three hundred watts, there is the problem of size and heat dissipation, and so separates are to be preferred.

Q. *How about receivers? Are separate tuners and amplifiers better?*

A. Much the same arguments apply here too. Two big problems with FM tuners using tubes was drift caused by heat dissipation, and, secondly, size. Now, many receivers have very elaborate amplifier sections and give a high standard of FM performance too. Even so, separate tuners are still recommended for those who want the last dB of performance. And, of course, they are even more versatile—why buy a new set of equipment if all you need is better FM reception?

Q. *I can see that transistors have many advantages over tubes—freedom from microphony, less hum, small size, and they do not age. But aren't these advantages realized at the expense of noise and distortion?*

A. Definitely not! Taking the question of noise first, modern audio transistors can have a significantly lower noise level than the very best tubes—which is one reason why they are extensively used in studio equipment and microphone preamplifiers working with extremely small signals. As for distortion, it must be said that many early solid-state amplifiers had a higher distortion than their tube counterparts. Crossover distortion was a problem at low volume levels and another factor was the actual *distribution* of spurious harmonics. Although the overall measurable distortion might be quite low, the proportion of high order harmonics, like the fifth, seventh, and ninth, was high. It has long been known that these high order harmonics are subjectively more unpleasant and in fact, several proposals have been made for a realistic "weighting factor." The effect, as far as the listener is concerned, is a harshness of the sound, variously interpreted at the time as a brilliance, a clarity, or "that transistor sound." But, of course, during the past few years, developments in circuitry and improvements in transistors have changed the picture completely. Modern solid-state amplifiers have a lower distortion and better signal-to-noise ratio than possible with tube amplifiers. Elimination of the output transformer (necessary with tubes) has meant that the designer can achieve stability, wide bandwidth, good transient response, and a damping factor effective over the whole band. True, all these parameters *can* be met with output transformers but not too easily and certainly not cheaply. For instance, to maintain the low frequency response, a high primary inductance is required, but the windings have to be sectionalized to reduce self-capacity or high frequency and stability will suffer. As a matter of interest, some time ago, Peter Walker demonstrated a bridge method of evaluating the distortion of an amplifier using any kind of input signal, *including speech and music.* Briefly, it involved the balancing out of the input signal with the output—what's left is distortion, a deviation from the original. It could be displayed on an oscilloscope or amplified and fed into a loudspeaker if so desired. Obviously, this method must be almost foolproof as it takes into account IM, THD, transient mutilations, frequency deviations, thermal effects, and so on. I said *almost* as phase effects could cause instability unless precautions are taken. At the demonstration, Peter Walker proved that the distortion detected by the bridge was greater on his old and respected Quad tube amplifier than on his 303 transistor amplifier!

Q. *If I buy an add-on power amplifier for quadraphonic sound, will I need the same power for the rear channels?*

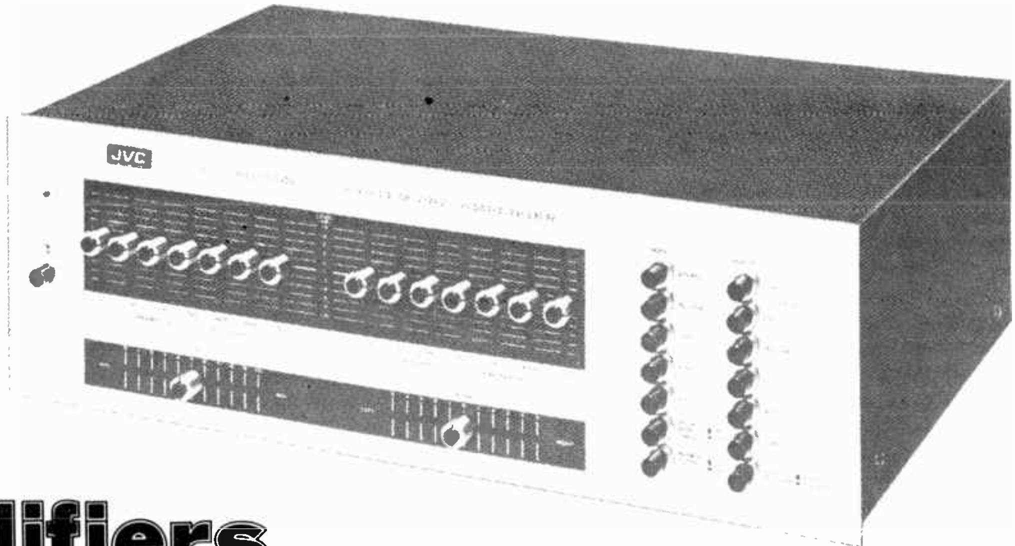
A. Yes. Many recordings demand equal power from all four speakers although a few use ambience only for the rear channels. *Overall* sound level, however, should be about the same as for conventional two-channel, although opinion is divided on this issue.

Q. *What noise level is inaudible? My amplifier is rated at 65 dB for* PHONO, *yet the background noise is quite loud.*

A. This is not an easy question to answer because there are so many factors involved. Here are some of them: Distribution of the noise (i.e., the proportions of low and high frequencies), loudspeaker characteristics, room characteristics, and method of measuring. A speaker system with a "peaky" treble will over-emphasize hiss, and obviously a low frequency hum of 60 Hz would be more audible from a loudspeaker with a resonance in that region. Again, a hum that would be completely inaudible on a small bookshelf speaker might be unbearably loud on a large corner horn system. In practice, a signal-to-noise ratio of 50 dB or above (referred to full amplifier rated output) would be unobjectionable.

Q. *Is there any point in buying an amplifier having a higher output than I need? Would it sound cleaner at low levels?*

A. In general, it is wise to allow as large a factor of safety as possible. It would surprise many people to know that what often passes as a roughness in the sound or is dismissed as a recording defect is actually overloading. Many of our present day loudspeakers are relatively inefficient (0.5% is not uncommon) and they really *do* need a fair amount of power, especially in a large, well-damped room. Usually, there is no detectable difference between very large and small amplifiers at low listening levels, although it must be said that transient peaks are often larger than many people imagine! A lot depends on the amplifier's overload characteristics—some clip peaks cleanly without fuss, while others produce an excruciating noise! *G.W.T.*



JVC Model 5011

Amplifiers

A look at requirements and specifications

George W. Tillett

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

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

Distortion

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

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

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

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

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

Intermodulation Distortion

This is the production of spurious sum-and-difference frequencies when two or more frequencies are passed through a non-linear amplifier. Figure 1(b) shows a 50 Hz and a 1000 Hz note applied to an amplifier which we will label B. It is a reasonably good amplifier and so the resultant waveform is fairly pure. However, if we apply the same frequencies to amplifier A, it is quite a different story. The distorted 50 Hz note deforms the 1000 Hz waveform by intermodulating it as shown in Fig. 1(c). In this process it creates a whole series of sum-and-difference frequencies—1050, 950, 1,100, and so on. These

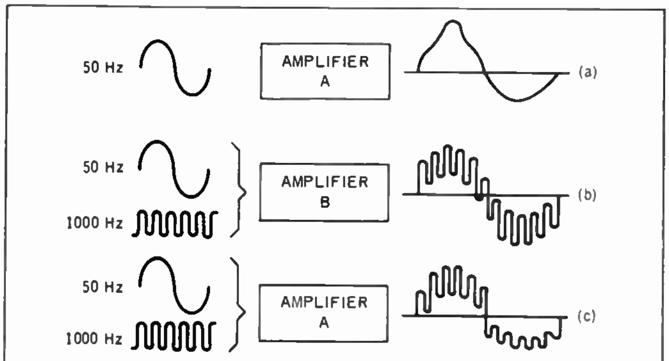


Fig. 1—Harmonic and intermodulation distortion.

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

Frequency Response

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

1. The internal bandwidth of an amplifier using a large

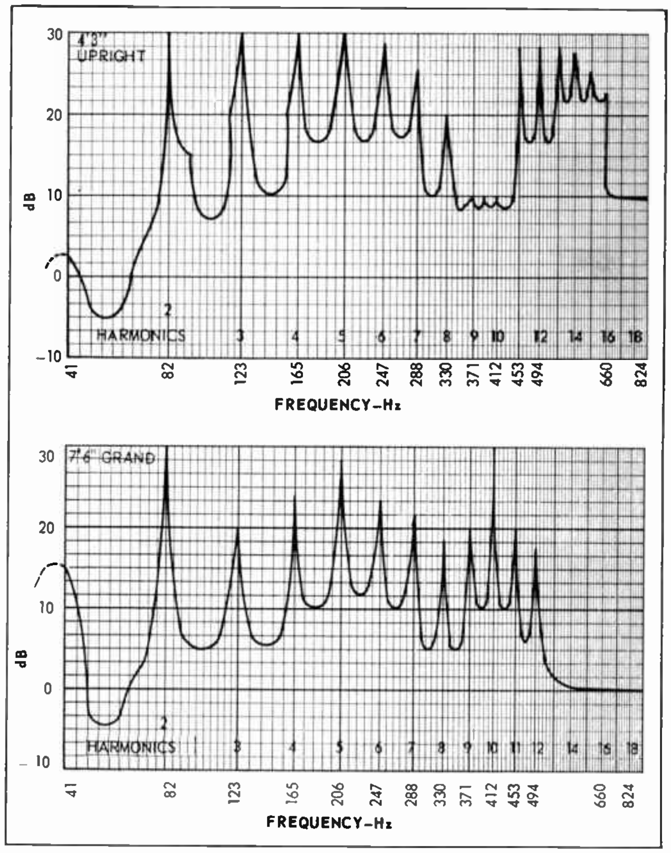


Fig. 2.—Harmonics of E3 (41.2 Hz) produced by two pianos using equal hammer velocity. (From "Pianos, Pianists, and Sonics" by G. A. Briggs, Cahners Publishing, 221 Columbus Ave., Boston, Mass. 02116.)

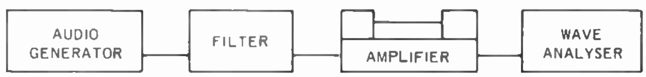


Fig. 3—Wave-analyzer method of testing amplifier distortion.

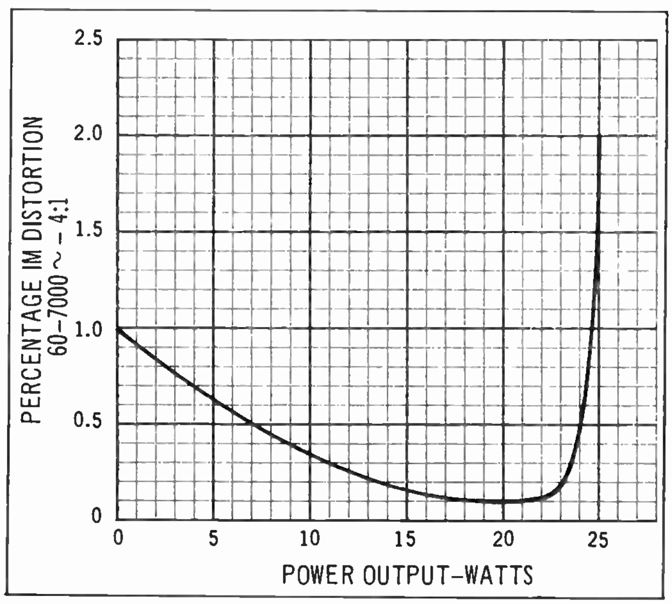


Fig. 4—Showing intermodulation distortion characteristic of a typical early transistor amplifier.

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

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

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

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

Power Output

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

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

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

That ± 1 dB Rating

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

Overload

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

Power Bandwidth

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

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

Damping Factor

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

Transient Response

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

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

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

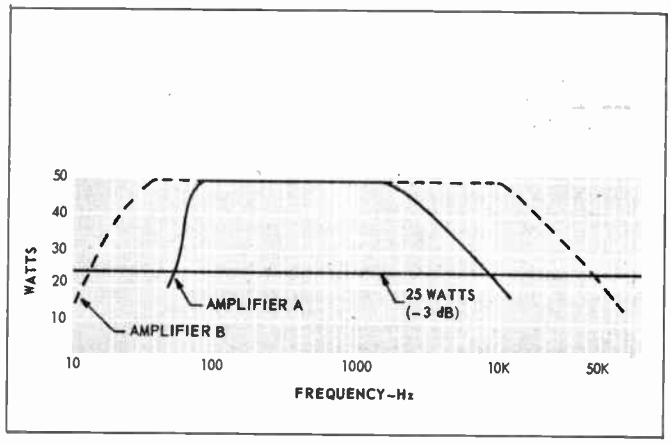


Fig. 5—Power bandwidth of two amplifiers.

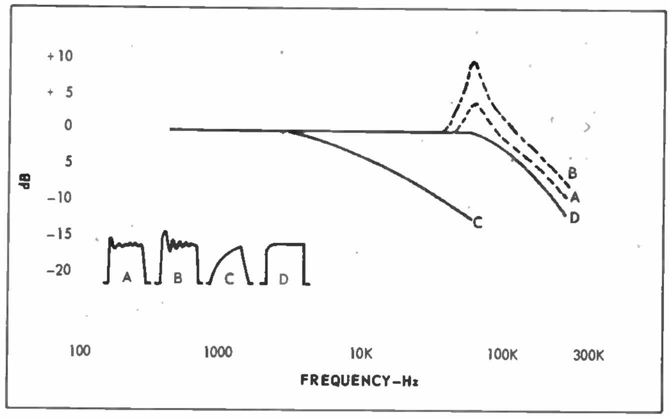


Fig. 6—Responses with 10 kHz square wave resolutions.

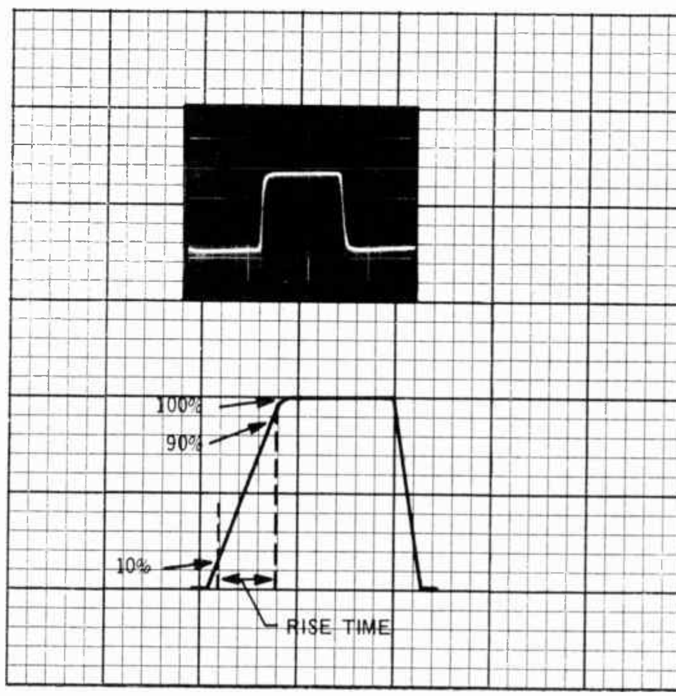


Fig. 7—The square wave rise time is the time taken to achieve 90 percent resolution from 10 percent.

Stability

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

Tone Controls

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

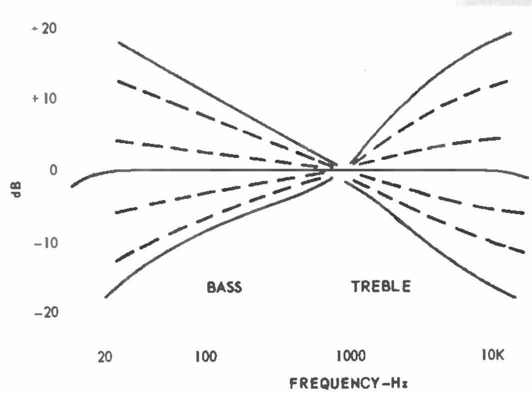


Fig. 8—Typical passive tone control characteristics.

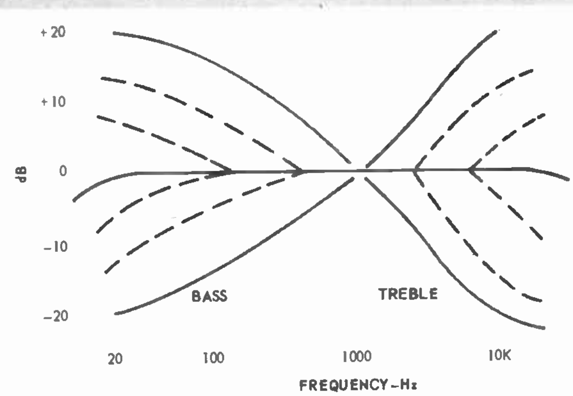


Fig. 9—Baxandall tone control characteristics.

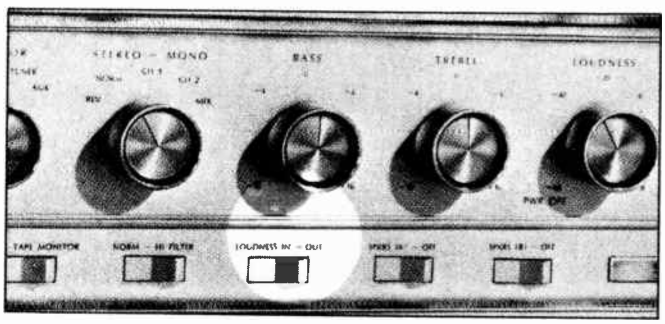
High Frequency Filter

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

Low Frequency Filter

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

Loudness Control



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

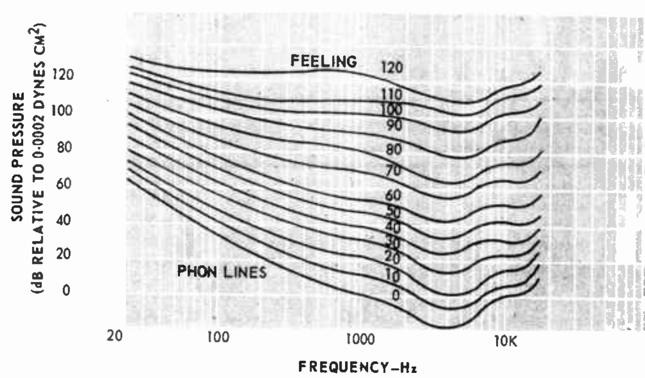


Fig. 10—Fletcher-Munson equal loudness curves.

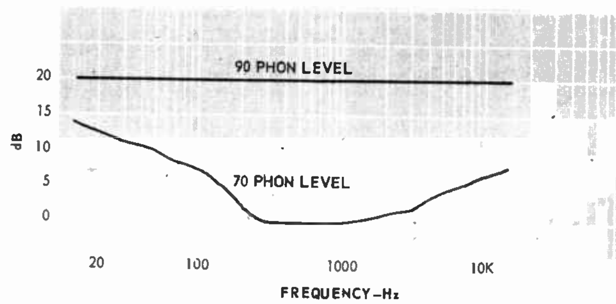


Fig. 11—Relative level curves.

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

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

Hum and Noise

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

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

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

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

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

Sensitivity

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

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

Channel Matching

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

Crosstalk

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

Facilities

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

Reliability

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

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

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

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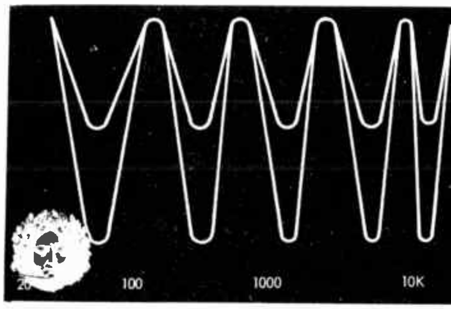
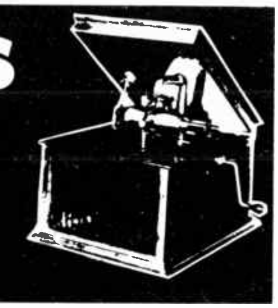


Buying Watts And Other Things



Leonard Feldman

Buying Watts and Other Things

ONCE YOU'VE gotten past the traumatic decision involving audio power requirements of your ultimate system, you'd think the choice of an amplifier (integrated, basic, or part of an all-in-one receiver) would be relatively simple. You've waded through the literature regarding power ratings. You now understand that continuous power ratings (often erroneously dubbed rms power) are more meaningful (and less inflated) than "dynamic power", "music power" or "IHF music power". You've learned to steer clear of products which feature "peak power" or even "instantaneous peak power" ratings. You've even selected your speaker systems and been told by the speaker manufacturer's brochures how much power you should supply to them (and, perhaps, how much power you'd better *not* supply—ratings in this area are still quite vague) and you're ready to shop for the electronic "heart" of your system-par-excellence. It's only then that you are faced with a host of *new* decisions.

Watts Versus Features

Basically, an audio amplifier has one primary raison d'etre. It's supposed to accept low-amplitude electrical signals which represent your various program sources and *amplify* them until they are powerful enough to drive your loudspeaker systems. It should perform this task without introducing distortion, and the broadest definition of distortion means the introduction of any differences to the output signal as compared with the input signal. That means differences in *content* (harmonic or intermodulation distortion) as well as differences in relative amplitude of the frequencies contained in the original signal (flat frequency response). Writers in this field have, at various times, suggested that the ideal amplifier is best described as a "piece of wire with gain" and not a few manufacturers have, over the years, used that cliché to describe their products.

In point of fact, though, as you begin to examine the amplifier products currently available, you find that this single objective is augmented by countless features including controls, switches, lights, jacks, sockets and other seemingly unrelated appurtenances which, at first glance, even seem to contradict the stated objective of "pure" amplification. It is these features that we'd like to sort out in terms of their usefulness (or superfluousness) in a high fidelity music reproduction system.

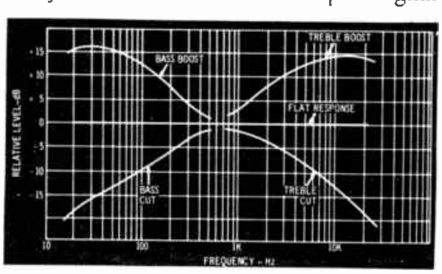


Fig. 1—Typical range of adjustment afforded by most tone control circuits.

Tone Controls

Only the arch-conservative purist will argue against the need for bass and treble tone controls in a modern hi-fi system. They're needed, we are told, to correct for all manner of sonic deficiencies which exist elsewhere in the system—such as poor lower bass response in our speaker systems, improperly equalized program source material, highly absorbent room furnishings (which gobble up the "highs") and the like. Yet, as you tour the audio shops (and the homes of your friends who own hi-fi setups), make a note of how few listeners actually move their bass or treble knobs away from the sacrosanct "flat" settings. There may be an ego problem here. To depart from "flat response" is to tacitly admit that "some other part of the system" is anything but flat and that implies poor judgment on the part of the audiophile in his "system assembly". On the other hand, the majority of "tone control" circuits may simply not be suited to the required "sound tailoring" job at hand. Consider Figure 1—the typical range of control afforded by the usual bass and treble controls. If, indeed, the deficiency noted in a particular system involves the need for boosting frequencies below, say, 150 Hz to make up for poor lower bass reproduction, the typical tone control cannot correct this situation without simultaneously boosting frequencies from about 500 Hz down, a solution which leads to so-called "barrelly" sound.

Selectable Crossover

Some amplifiers switch positions (two, three or even more) which predetermine at what frequency boost or attenuation begins when the tone controls themselves are rotated. Thus, as shown in Fig. 2, it would be possible with such an arrangement to set up a response curve which only emphasizes (or attenuates) the extremes of frequency which require such alteration.

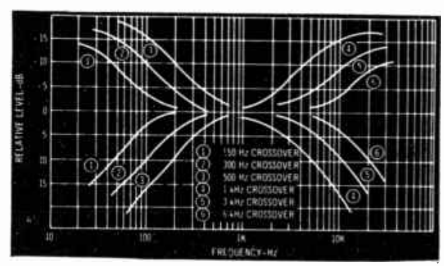


Fig. 2—Selectable crossover tone controls offer greater choice of tonal correction possibilities.

A lovely feature *if you need it*. Obviously, if your tendency is to leave the tone controls in their flat position (or even by-pass them by means of a suitable switch sometimes provided for that purpose), then the added cost of selectable cross-over tone control circuits is not for you.

Graphic Equalizers

On the other hand, if your listening acuity is such that even selectable crossover tone controls fail to adjust the response to what you consider to be correct, you may want to consider an amplifier with *more* than the usual bass and treble controls. The first step in this direction is to be found on several models which now offer mid-range tone controls, as well as bass

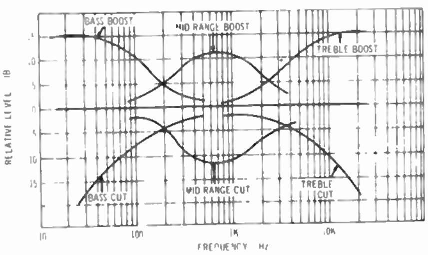


Fig. 3—Mid-range tone control permits additional response alteration.

and treble. As shown in Fig. 3, this extra control offers adjustment of mid-frequency emphasis. Early versions of this feature used to be called "presence" controls, in that they tended to emphasize the sound of vocalists—since vocal programming is primarily in the mid-audio range.

If your acoustic environment is such that even *three* tone controls won't suffice, there are amplifiers on the market that sport as many as *five* separate tone controls, each able to control a specific segment of the audio frequency spectrum. The name "graphic equalizer" has been applied to such multiple tone control arrangements and, if self-contained segmented tone controls still don't satisfy you, you can purchase *separate* graphic equalizers with ten, twelve or even twenty-four segmented controls. The action of a five-segment graphic equalizer built into an integrated amplifier is shown,

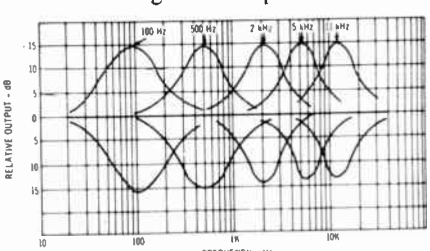


Fig. 4—A five-segment tonal equalizer provides even more accurate tailoring of overall frequency response.

graphically in Fig. 4. Obviously, the more segments—the more circuitry, and the more circuitry, the higher the cost.

Tape Monitor Facilities

Speaking of "add on" devices to your amplifier, manufacturers of such devices ought to be eternally grateful to some remote tape-recorder manufacturer of yesteryear who had the bright idea of building separate "record" and "playback" heads into his machine, plus enough electronics for both to operate simultaneously. This clever innovation permitted the recordist to "monitor" the results of his recording efforts a fraction of a second after the tape has been recorded—providing he could feed the output of his tape-deck's playback preamplifier into a suitable input on his amplifier—the *same* amplifier he was using as a program source for *making* the recording. In order to do this, amplifier manufacturers provided a switch called "Tape monitor", which is nothing more than a means for "breaking" the signal path at a suitable point in the amplification chain. The program to be recorded is fed to the tape-deck from the "source" side of the "break", while the resultant recorded signal is fed to the other side of the "circuit break"—the side that ultimately leads to the loudspeakers or output of the amplifier. Once this "break" became universally available on most component amplifiers, it also served as the necessary connection point of all manner of devices (including the graphic equalizers that are bought separately) which could be "added" to a system. Were it not for this simple circuit-interruption point, it would be impossible, for example, to add any of the four-channel decoders which now permit easy conversion of stereo systems to quadraphonic sound.

Some amplifiers now feature two or even three tape-monitor jack-pairs. Among other things, this permits the user to record onto two tape recorders simultaneously or to dub from one recorder to another. Again, if you are *not* that heavily involved in recording work, the presence of two or more tape monitor circuits is a redundancy you should not have to pay for.

Phono Inputs

Speaking of "doubles", many amplifiers offer multiple pairs of phono-input jacks. In some, both pairs are identical and offer equal sensitivity, regardless of which pair is used. These arrangements are intended for the owner of, say, a record changer and a manual turntable who might want to do casual, extended listening via his record changer but may want to "single play" more critical recordings on a manual turntable. In yet another arrangement, some amplifiers are equipped with pairs of phono inputs which have different sensitivities. For example, the PHONO 1 inputs may be designed for cartridge outputs in the range of from 1 to 3 millivolts, while the PHONO 2 inputs may accept cartridges having outputs from 4 to 8 millivolts. Obviously, if you *know* what the nominal output of your phono cartridge is and do not plan to own two record playing facilities, there's really no point in spending the extra money for the dual phono input feature.

Filters, High and Low

The use of filters for the "elimination" of turntable rumble and high frequency noise (record scratch, tape hiss, FM background noise) is, at best, a compromise remedy. There's no question about it—filters *do* alter what is often "perfectly flat" frequency response. Rumble consists of very low frequency signals (usually below 60 Hz) and in order to reduce its effects, an amplifier's response must be attenuated at very low frequencies. Unlike "bass tone control action", however, such filters are designed to start cutting at or about the frequencies which are

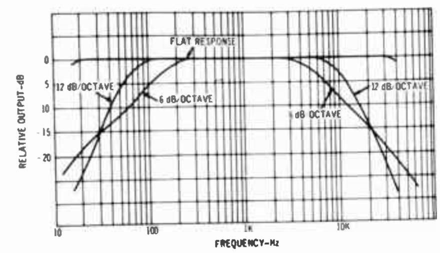


Fig. 5—Filters attempt to cut out unwanted rumble and hiss without destroying too much of musical value.

involved. This is illustrated in Fig. 5. The action of two kinds of low frequency filters is shown. Note, that the more gradual sloping response curve has to start cutting frequencies at about 150 Hz in order to provide 15 dB of attenuation at 30 Hz. This filter has a slope of 6 dB per octave. The preferred 12 dB per octave filter (steeper sloped line) needn't start altering frequencies above 100 Hz to afford the same 30 Hz attenuation. These two values of "slope" are generally found in high-frequency filters, as well. Both filter types must necessarily bite into musical content, and if you don't suffer from turntable rumble or record scratch hiss, is there any point in "buying" filters as part of your amplifier arrangement?

Microphones and Mixing

The breach between "professional" amplifying equipment (normally associated with recording studio or broadcast use) and "consumer" equipment continues to narrow and quite a few amplifiers sold for home use now include microphone input jacks. Some of these simply avail themselves of the high amplification capability of the phono preamplifier stages—removing the required RIAA record equalization and connecting the microphone input jack to these circuits by means of a position on the selector switch. This simple arrangement really offers very little more flexibility than the microphone inputs already available on your tape recorder, unless you take special delight in using your entire high fidelity system as a public address system in your living room. There are, we are told, people who suffer from an affliction which is the reverse of "mic fright". They love to hear themselves over loudspeakers—it seems to provide the extra sonority akin to that special sound one hears when singing in a tiled shower. If you get your kicks that way, than the extra mic position on your amplifier's selector switch will provide that extra ego stimulant at very little extra cost.

If, on the other hand, you want to "mix" live sounds with other program sources, the amplifier you choose should be equipped with a separately controlled microphone preamplifier circuit—one which is capable of being mixed in with other program sources and one which has its own level control, apart from the master volume control, so that you can attempt that "professional" mix of program" just like the recording engineers do". If you're really serious about "live recording", however, you may be better off using one of the many "outboard" microphone mixers which not only contain preamplification but provision for four or six microphones to be connected—each with its own associated level control. Outputs of most of these mixers plug simply into a high-level (Aux) input on your amplifier or receiver.

Input Level Adjustment

While a manufacturer has total control over the audio level of AM and FM radio signals recovered in his receiver product, he cannot predict what levels of phono signal, tape signal and the like you're likely to feed into the other various inputs of his product. If you pick products that all produce voltage outputs that are about equal and also equal to the internally supplied AM or FM detected audio signals, you're in luck. When you switch from one program source to another, you won't have to race to the volume control to adjust for differences. More than likely, however, one or more of your signal sources is going to be greater in amplitude than the others and when switching from or to that source, it can be quite annoying (if not disturbing) to experience extreme level shifts. One of the ultimate refinements offered by some receiver manufacturers and quite a few amplifier makers consists of a group of input level controls, each associated with a particular pair of input jacks. These enable you to carefully adjust all program source voltages so that as you switch from one program source to another, loudness remains constant.

The idea is lovely and of course, it does cost money in terms of extra controls on the amplifier or receiver. Allowing, even, that you are that much of a perfectionist, there is still room for redundancy. For example, if your tape player has its *own* output level control, you can set it to correspond with radio levels. If you own a separate amplifier and tuner, the tuner may well have an output level control too, obviating its need on the input of the amplifier. About the only instance in which separate input level controlling is not likely to be available other than on the amplifier is that associated with phono listening. However, as we've already pointed out, many amplifiers offer a choice of input sensitivity here, so you may not encounter too great a level shift even in the case of records.

For The Amplifier That Has Everything

The remaining few niceties that we'd like to mention fall into the category of "luxury" features that really don't contribute audibly to the performance of an amplifier but are, nevertheless, available as "convenience" features. We will not comment upon their "usefulness' but simply mention that they can be had—all at added cost since all involve additional circuitry and/or mechanical parts additions.

Included in this group are such things as attenuator switches, program indicator lights and level meters. The attenuator switch, which may not be familiar to most readers, is simply a switch which, when thrown, reduces overall listening level by about 20 dB. It's supposed to be used when the phone rings and you want to lower listening level so that you can hear the party on the other end of the line. Obviously, you could turn down the volume, but the theory here is that by throwing the switch during the phone conversation you can restore *exactly* the same listening level you had before after the phone is hung up. We leave that one to your own judgment.

Program indicator lights are great fun to watch as you rotate your program selector switch or push the selector buttons. Unless your amplifier is equipped with a remote control cable with which you can switch programs from many feet away, we fail to see what information they add in operating the amplifier, since you can read the designations around the selector knob just as easily at the time of program selection. Of course, some people forget easily, I suppose, by the time they reach the comfort of the listening chair or sofa, so if it's lights you want, lights you shall have.

Recently, level meters have found their way onto home amplifiers. Often, they provide a good indication of how much power is being supplied to the loudspeakers and, with today's more powerful amplifiers, this often serves as a warning device if the meter calibration is accurate. It is not rare to "blow up" or destroy a speaker by feeding it too much power and if you watch the meters, this tragedy can sometimes be prevented. If, however, the meters are used to establish correct recording levels, it must be fairly stated that most recorders (even cassettes) have more accurately calibrated record-level meters for this purpose, so once again, the presence of meters on an amplifier for this purpose may be a useless redundancy.

It has not been our purpose to put down any of the modern features and controls associated with the current crop of solid-state amplifiers. On the contrary, we think that manufacturers have brought a great deal of ingenuity and innovation to their products. Receivers and amplifiers sold for home use today often exceed the performance capabilities and control flexibility of the professional equipment of just a few years ago. All we're saying is that the prospective purchaser of a home music system should try to evaluate his needs before taking the plunge and, if possible, purchase the correct amount of power *and* the appropriate features which he is likely to use in the foreseeable future. Admittedly, as one goes up the scale in power and price, one generally finds more and more of the features we've discussed, so that it's often difficult to separate the "features" from the power rating and low-distortion capability of some products. There are exceptions, however. *Some* low-power units offer very low distortion plus a host of control features such as those we've discussed. Some super-power units offer a minimum of extra controls and switches and a maximum of good audio performance. You're likely to find just about everything you want in between if you shop carefully and knowledgeably. The point of the whole story is: but anything you need—but *use* what you buy if you really intend to get your money's worth.

Intermodulation Distortion:

A Powerful Tool for Evaluating Modern Audio Amplifiers

Gerald Stanley & David McLaughlin

THIS ARTICLE has been written to suggest a more helpful way of reporting amplifier distortion specifications to the prospective buyer. The comparative merits of two types of distortion testing will be presented, leading to the conclusion that one test shows some clear advantage over the other.

The widespread appeal of high fidelity has been strengthened by the broad range of equipment currently available. The increasing variety is an undisguised blessing since it allows every enthusiast the freedom to satisfy personal tastes in building his own system. Unfortunately though, the process of purchasing a system becomes more complicated with this diversity. The expense involved and the individual nature of a high fidelity system usually results in a great deal of evaluating and comparison by the careful customer, who naturally wants good performance for good money. Any performance information that can make the evaluation easier will be very beneficial.

Obviously, the effective communication of technical performance data is not an easy thing. First of all the information has to be presented in terms that the buyer can easily handle. After all, why should it be necessary for someone to have an intimate knowledge of audio electronics in order to make an intelligent purchase? It is also important that standard terms be used by all manufacturers in order to facilitate comparisons between different brands of equipment. Finally, since knowledge and equipment fall short of perfection, a constant updating of test procedures becomes essential. Unfortunately for the customer, the information he needs does not come from all manufacturers in a standard form, and traditional rating methods give way very slowly to more effective practices. This is not to say that current performance specifications are not helpful in making buying decisions, but to point out that they could be more helpful.

One of the most often quoted and inadequately defined technical specifications in the audio field involves the distortion produced in audio amplifiers. The term "distortion" covers a multitude of audio evils. Basically it describes a change in the original signal introduced by the electronic and mechanical equipment employed in reproducing the signal. An example is drawn in Fig. 1. The goal of any audio reproduction system is to approach to a greater or lesser degree (for a greater or lesser number of dollars) a perfect duplication of an original production of a piece of music. The sound of the original performance, whether it comes from Van Cliburn or Flatt and Scruggs, becomes the standard. Any changes in the original sound can be described as some form of distortion.[1] In the process of translating an original performance onto a disc or tape, a certain amount of distortion is introduced. When the record or tape is played, the high fidelity reproducing system introduces other subtle changes and the resulting sound moves a little further from the original. As you would expect, the degree of change or distortion depends on the overall quality of all equipment that has been used in the process. Interestingly enough, the distortion may not even sound unpleasant, but as long as it represents a difference from the original, it is distortion by definition. To minimize overall distortion, then, it is generally important to have each piece of equipment in a system[2] produce a minimum amount of distortion. This seems rather obvious, but because of the difficulty of comparing distortion specifications which are stated differently, it is not always easy to determine which equipment will produce the least objectionable distortion.

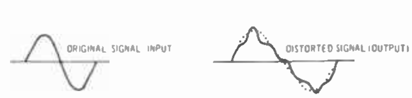


Fig. 1—Example of distortion

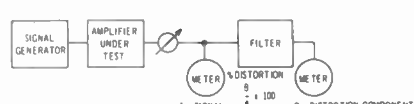


Fig. 2—Harmonic distortion test setup.

Two basic methods have come into use for measuring distortion in audio amplifiers. Unfortunately the results of one method do not necessarily indicate the results of the other. To compound the problem, the methods are used with varying degrees of thoroughness, which give varying amounts of useful information about the performance involved. Simply stated, the truth is that the buyer is not being helped as much as he could be.

To begin with, the problems involved in evaluating amplifiers are rather significant. For instance, distortion tests are not performed while the amplifier is actually handling music. (See "Amplifier Q's and A's—Mainly for Beginners" in this issue.) Of course, the equipment will actually be used to play music, but the situation is too complex to permit a practical distortion test to be performed. Therefore, the actual testing requires some simplification. The test conditions must also be standardized, so they can be duplicated accurately by anyone wishing to check the results. Supply voltages, signal level test equipment capabilities, loads and so forth must be specified wherever they have a significant effect on the outcome of the test. When these variables are defined, more meaningful comparisons can be made between different brands of equipment. The test should also illuminate thoroughly the particular qualities (good and bad) of the equip-

Crown International, Elkhart, Indiana

ment being tested. For instance, a test could be made which would show excellent performance at a particular operating point of an amplifier while completely ignoring the performance at other equally important operating levels. To summarize, a test should be a simplified and repeatable version of actual operating conditions, which adequately covers the range of performance expected. Both of the methods commonly used to evaluate audio amplifier distortion are simplified, repeatable versions of actual operating conditions, which can be used to check the whole range of expected performance. One, however, offers particular advantages in the evaluation of modern audio amplifiers.

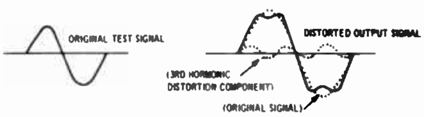


Fig. 3—Example of harmonic distortion.

First, let us consider the traditional method—harmonic distortion testing. This involves evaluating an amplifier's performance as it handles a one-frequency signal. The complex musical signal is thus approximated by a single frequency. The test signal must be as free of distortion as possible, so that its inherent distortion is not confused with the distortion introduced by the amplifier. Essentially, the test signal is passed through the amplifier and the resulting output signal is checked for changes from the input. The general arrangement of equipment is shown in Fig. 2. The output signal is measured, after which the original test frequency is filtered out. The remaining output signal measured, on the assumption that what is left over consists of unwanted distortion components added by the amplifier. As typically produced (ignoring effects of hum and noise), this distortion is called harmonic distortion because the unwanted additions to the single tone can be separated into the harmonics of that tone. By way of illustration, if 440Hz is used for the test signal, the second harmonic will be 880Hz, the third harmonic will be 1320Hz, and so on. In this case, harmonic distortion measurements should indicate the prominence of these distortion frequencies: 880Hz, 1320Hz, etc. Figure 3 gives an example of the appearance of harmonic distortion on an oscilloscope display. Figure 4a shows where the harmonics appear on a frequency spectrum. Some idea of the relative sounds involved can be gained by sitting down at a piano and sounding middle A (440Hz). The first harmonic (880Hz) would be A one octave higher. The next octave would produce the third harmonic (1320Hz) and so forth. Since these are all in harmony, playing them together will obviously not produce an unpleasant sound, but the sound will definitely be different from the sound of middle A alone. In the actual test, a wave analyzer may be used to look at each of the harmonics individually, to see how much each contributes to the total distortion. In some equipment, the second harmonic may be the largest component, while in other equipment the third or some higher harmonic may contribute the most. Usually a total harmonic distortion (THD) figure is stated, which ideally expresses the rms sum of all the harmonic distortion components together as a percentage of the rms fundamental signal.[3] For the sake of thoroughness, the tests should be repeated at different frequencies and at different power levels, although this takes much more time and effort.

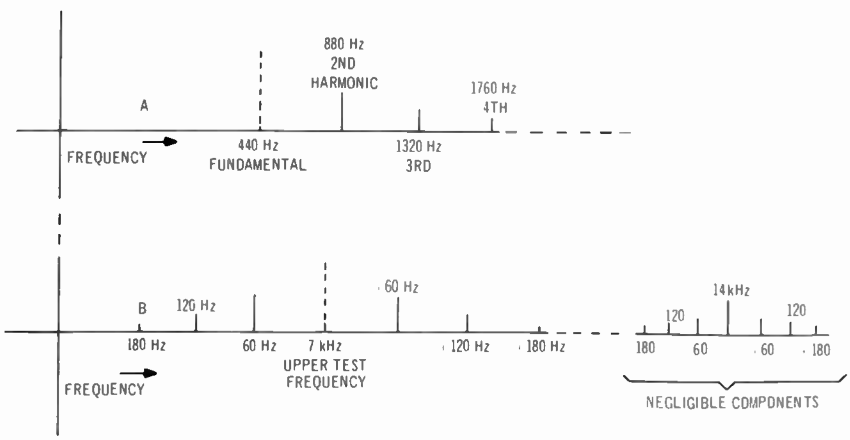


Fig. 4—A, Harmonic distortion components, **B**, intermodulation distortion components.

Fig. 5—Example of IM distortion

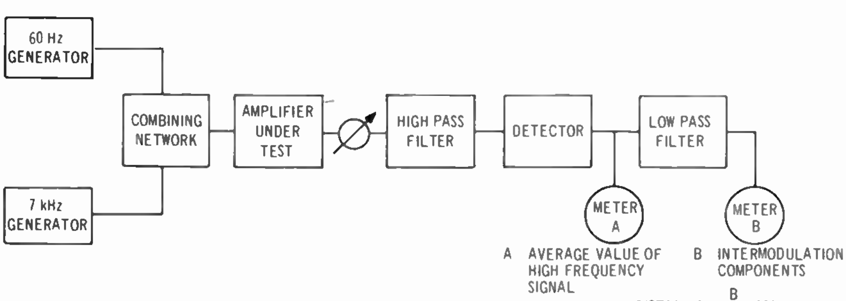


Fig. 6—IM distortion test setup.

The second and acoustically more relevent method of distortion testing measures intermodulation (IM) distortion. This type of test evaluates amplifier performance as it handles a two-frequency signal. The complexity of a musical signal is again simplified, this time being approximated by the interactions of two frequencies. As defined by the Society of Motion Picture and Television Engineers (SMPTE), the IM distortion test signal is made up of two frequencies in a 4:1 amplitude ratio of low frequency to high frequency. Typically the two frequencies are 60Hz and 7KHZ (Fig. 5). The general test arrangement is shown in Fig. 6. The output from the amplifier passes through a filter which removes the low frequency (60Hz) test signal. The remaining output, consisting of the high frequency test signal (7KHZ) plus the distortion modulation components, is AM detected[1], after which everything but the distortion products is removed by a second filter. The distortion modulation components are measured and expressed as a percentage of the total AM detected signal. The primary distortion measured comes from the interaction of the two test frequencies. The 60Hz frequency will modulate the

7KHZ frequency and form sum-and-difference frequencies, such as the sum of the two (7060Hz) and the difference between the two (6940Hz). Other sum-and-difference frequencies will also appear involving the harmonics of both frequencies. Figure 4b shows what some of these will be on the frequency spectrum. For the purpose of practical measurement, only the distortion components around 7KHZ are significantly large and these are the ones measured as distortion. Figure 5 shows how IM distortion of the test signals might appear on an oscilloscope display. Again using the piano to illustrate, an idea of the kind of sound involved here can be gained by sounding middle A again, and then sounding middle A along with the white keys on either side of it (G and B). These two notes are between 50Hz and 60Hz different from A, and when played together with A, demonstrate the kind of dissonance resulting from intermodulation distortion.

Depending upon the particular conditions of the test, such as the characteristics of the equipment being tested, the frequencies used may be changed and the 4:1 amplitude ratio between frequencies may vary. Generally the 4:1 ratio of 60 Hz and 7KHZ is used because it provides a realistic example of the musical situations for which an audio amplifier is designed. IM tests should be run at a wide range of output power levels to reveal problems that may show up only at particular levels. As an example which will be discussed in more detail later on, IM testing shows excellent sensitivity to low power crossover notch distortion, which is a traditional sore spot of some solid state amplifier designs.

Now that we have briefly discussed both methods, you might naturally ask how they are related, but this is not a simple nor brief proposition. A great deal of discussion has been published[5] with impressive mathematical support to describe this elusive relationship, but the results do not apply to most equipment. Several common (and sometimes desirable) characteristics of electronic equipment can each or all remove any predictable relationship between IM and harmonic distortion. At a given peak power level, and within the normal operating range of high fidelity amplifiers, IM distortion typically runs from two to six times as high as harmonic distortion. In any individual case, however, it is necessary to run both tests if both harmonic and IM figures are needed. This lack of a simple means by which to compare IM and

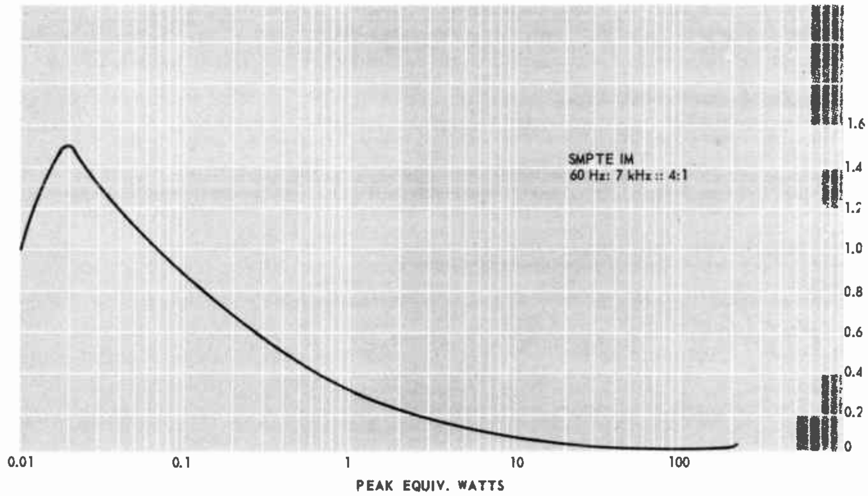


Fig. 7—IM evidence of crossover notch at low power.

harmonic ratings suggests that the customer would prosper if one method were consistently used, in which case he could make meaningful comparisons. For a number of reasons, IM testing is the logical choice.

To begin, there are significant weaknesses in the harmonic testing procedure. First, the harmonics detected as distortion are not always offensive to the listener. The piano experiment suggested above should illustrate this, along with the fact that musical sounds are frequently made up of harmonic combinations. Second, the single-frequency test signal does not resemble typical program material and the results do not indicate the kind of complex interactions that occur between different frequencies. This can result in ignorance of serious deficiencies in the equipment. Third, the usual THD figure groups all harmonic components together, which can mask the fact that most of the potentially offensive distortion comes from high order (higher frequency) harmonics[6]. An amplifier generating mostly high order distortion products may then sound worse than another with the same THD rating which produces lower order distortion. Fourth, THD measurements group noise along with harmonic components and thus may produce a mischaracterization of a product. Fifth, harmonic distortion testing instruments may have residual distortion levels above the distortion levels of the amplifiers under test. It is difficult to inexpensively produce and analyze a test signal with distortion lower than state-of-the-art audio amplifiers. Sixth, the test procedure is unwieldy. In the usual process, some fine tuning is involved to completely remove the test signal before the harmonics are measured, a procedure which needs to be repeated at different frequencies, and then at different power levels for each frequency. This results in a sensitive operation being performed many times for a single piece of equipment. Many of the aforementioned weaknesses could be lessened by the use of a wave analyzer, but this would not help the problems of expense and time involved.

In contrast, IM testing offers clear advantages over harmonic testing. First, the sum-and-difference frequencies detected as distortion by IM testing are not harmonically related to the original signals and therefore constitute a much more audibly obnoxious type of distortion (as suggested by the piano experiment). Second, the use of a two-frequency test signal provides a simple but more realistic approximation of musical material, and the test results indicate the interactions between frequencies that can be expected in actual use. Third, the use of the 4:1 SMPTE amplitude ratio gives an inherent prominence to more audible high-order distortion products, which in turn brings about better agreement of IM test results with listening tests[7]. Fourth, SMPTE IM measurements concentrate on a relatively narrow band of frequencies around the upper test frequency, a situation which serves to keep hum and other noise out of the final test results. Fifth, it is possible to obtain reasonably-priced IM distortion measuring equipment with residual distortion levels below those of state-of-the art audio amplifiers. Sixth, since there is no tuning needed to filter out the test frequencies and since two frequencies in combination provide a test for the entire audio bandwidth. the only change necessary during the test is in the power level. With proper equipment, IM testing can thus be done very quickly and efficiently.

Despite these advantages, IM distortion testing has found limited use and has sometimes been used to poor advantage. It is most important to cover an adequate range of power levels if an amplifier is to be thoroughly tested. As mentioned before, crossover notch distortion has plagued many solid state amplifiers. Fortunately, this type of distortion generates high order terms which quickly show up in SMPTE IM measurements *if* the tests are made at the levels (as low as 10 milliwatts) where crossover problems occur. Testing down to a level of 1 watt (a commonly used lower test limit which is frequently understood by the expression "all power levels below rated output") hardly ever reveals the cross-over notch distortion (e.g. In a 100 watt amplifier this is only 20dB below full output whereas music may cover 70dB.) Figure 7 illustrates the kind of IM increase that can occur at low power levels.

To summarize, harmonic distortion testing, on the surface is very simple conceptually and can be useful in equipment for which SMPTE IM testing would be inadequate (such as a graphic equalizer where low and high frequencies follow different signal paths.) But for many situations and in particular the case of audio amplifier testing, IM distortion measurements offer distinct advantages both to the manufacturer and to the consumer. Due to the simplicity of such tests with a modern, inexpensive IM analyzer, serious customers should insist on a fully documented plot of IM distortion versus output power, a request which quality manufacturers will happily fulfill. Æ

1. A separate problem is noise, which involves the addition of unwanted sounds not related to the sound being reproduced, such as hum from power supplies, etc. Important kinds of distortion included in the definition above, but not considered in this discussion, are phase distortion and amplitude distortion. Phase distortion deals with the shifting of the complex relationships among the different tones of a musical signal and is generally much more subtle than harmonic or intermodulation distortion. Amplitude distortion results from variation of gain with frequency and shows up as poor frequency response.

2. This assumes that the equipment has been chosen so that the different components will be compatible with each other, or noninteractive. (For example, damping factor is a measure of noninteractiveness of amplifiers and loudspeakers.) Otherwise, some part of the system will be improperly loaded or driven and distortion will occur regardless of the quality of the equipment.

3. Commercial THD analyzers actually measure the average of the distortion signals taken as a percentage of the average distorted amplifier output, rather than measuring rms figures.

4. This operation in effect demodulates the high frequency signal from the high pass filter. From the demodulated signal an average value is taken as a reference for the percent distortion. The intermodulation components (low frequencies) are then separated from the high frequency by the low pass filter.

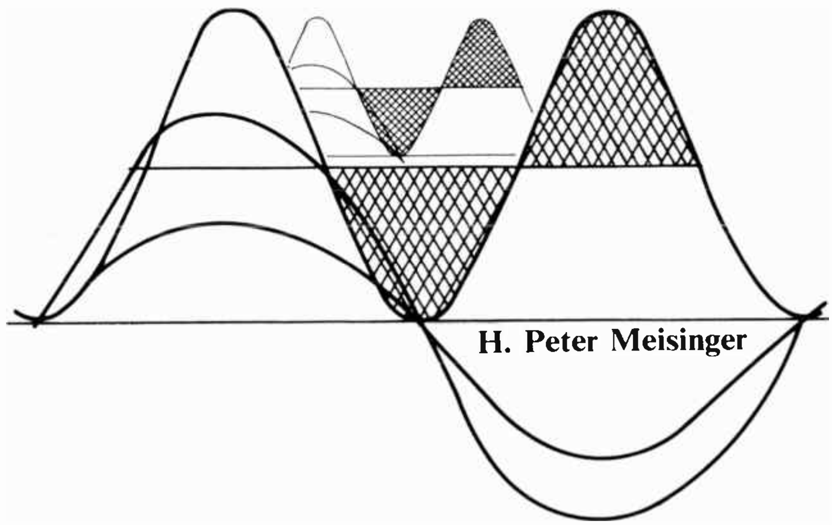
5. See, for example, Callendar, M.V. and S. Matthews "Relations between Amplitudes of Harmonics and Intermodulation Frequencies," *Electronic Engineering* (June, 1951), P. 230, and D.E. O'N. Waddington "Intermodulation Distortion Measurements," *Journal of Audio Engineering Society* (July, 1964), Vol. 12, #3, P. 221.

6. Shorter, D.E.L. "The Influence of High Order Products in Non-linear Distortion," *Electronic Engineering* (April, 1950), P. 152.

7. Callendar and Matthews, *op. cit.*

8. See also "Amplifier Requirements & Specifications," AUDIO, (April, 1971), Vol. 54, #4, p. 32.

What's Watts



H. Peter Meisinger

In the early days of hi-fidelity, amplifier power ratings were quoted in watts. To the early hi-fi component manufacturer, the advertised watts in reality meant average watts. As hi-fidelity became more popular, a myriad of new inflated power ratings began to appear. (e.g. music power, peak power, etc.). In a seeming attempt to clarify power ratings, "the rms watt" has come into recent usage.

Recently, a great deal of discussion has taken place regarding the use of the term rms watts (1). The industry, in general, has adopted the use of the terms rms power, or rms watts. Many of the well-known testing laboratories also use these terms. Unfortunately, the terms rms watts, and rms power are incorrectly used.

In the laboratory, amplifier power is determined by measuring the voltage across a calibrated resistor with an rms voltmeter. This would lead to the seemingly logical conclusion that the watts determined in this manner would be rms watts. This is simply not true.

Ammeters and voltmeters that are used to measure alternating currents or alternating voltages invariably are calibrated in terms of the rms value of a sine wave unless otherwise specified. However,

1. The product of rms volts times rms amperes yields *average watts*, not rms watts.
2. The product of rms amperes squared times the circuit resistance yields *average watts, not rms watts.*
3. rms voltage squared divided by the circuit resistance yields *average watts, not rms watts.*

First, let us deal with voltage and current (amperes). The circuit shown in Figure 1 indicates an alternating current sine wave generator whose peak voltage is determined by the oscilloscope to be one volt. The generator is loaded by a one ohm resistor. Since current (amps) = $\frac{\text{volts}}{\text{ohms}}$, the numerical value of the current will be the same as the numerical value of the voltage since the divisor is equal to one (ohm).

Figure 2 illustrates the instantaneous current (I), instantaneous voltage (E) and instantaneous power in a resistive circuit.

The peak power is equal to the product of the peak voltage and the peak current.

$$P\ \text{peak} = E\ \text{peak}\ I\ \text{peak}$$

Similarly, the power at any instant in time is equal to the product of the instantaneous voltage and instantaneous current.

$$W\ \text{inst.} = E\ \text{inst.}\ I\ \text{inst.}$$

The instantaneous power curve of Figure 2 is this product, and is seen to be a sine wave of double frequency without negative values.

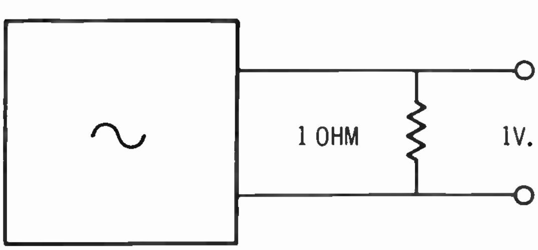


Fig. 1—indicates an alternating current sine wave generator whose-peak voltage is one volt.

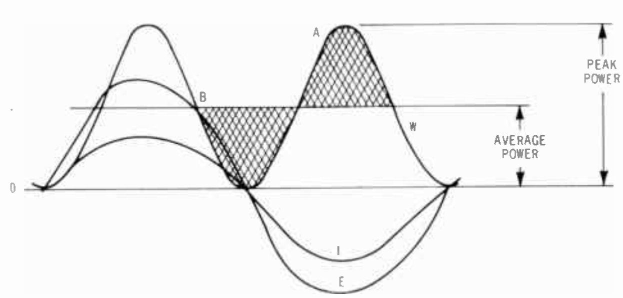


Fig. 2—illustrates the instantaneous current voltage and power in a resistive circuit.

(1) J.R. Ashley "What's a Watt (rms)?" *J. Audio Eng. Soc. Vol. 19, p. 793 (Oct. 1971)*

J. G. McKnight comments on "What's a Watt (rms)?" *J. Audio Eng. Soc. Vol. 20, p. 46 (Jan-Feb 1972)*

John Eargle and Bart Locanthi "RMS Power: Facts or Fancy" *J. Audio Eng. Soc. Vol. 20, p. 45, (Jan-Feb 1972)*

A line drawn equidistant between the peaks and valleys of the power curve represents the average power and is seen to be equal to fifty percent of the instantaneous peak power. This is graphically illustrated by observing that the shaded top section of the sine wave fits perfectly into the shaded adjacent valley. This represents the average power.

Since the average power line is drawn at the mid-point of the power curve,

Average power =

$$\frac{\text{Peak voltage} \times \text{peak current}}{2} = \frac{\text{E peak}}{\sqrt{2}} \times \frac{\text{I peak}}{\sqrt{2}}$$

In figure 1:

$$\frac{1}{\sqrt{2}} \times \frac{1}{\sqrt{2}} = .707 \times .707 = 0.5 \text{ watts average power}$$

$$\text{I rms} = \frac{\text{I peak}}{\sqrt{2}} = \frac{1}{\sqrt{2}} = \frac{1}{1.4} = 0.707 \text{ of the peak value}$$

Therefore, the rms value of voltage or current equals .707 of the peak value, or conversely, the peak value equals 1.414 times the rms value.

NOTE: This is true for voltage and current, not power.

In dealing with voltage and current, the terms "rms" and "effective" are used interchangeably. The real significance of *rms* (or *effective*) voltage and current is the fact that it will cause the same amount of heat to be generated in a resistance as a numerically equal d.c. source. In other words, *average* sine wave a.c. watts generate the same amount of heat in a resistance as a numerically equal d.c. source. The watt-hour meter in your home measures average watts or heating effect, not rms watts.

rms, when used with voltage or current, permits $W = I^2 R$ and $W = \frac{E^2}{R}$ to be used alternately between direct current and sine wave a.c. since the heating effect is the same in either case.

An additional proof of the above was rather simply done using one of the new scientific pocket calculators. All computations were made to ten places, however, with the limited number of sampling points only three place accuracy can generally be assumed.

Table 1 shows the calculation of average and rms current and voltage and average and rms watts.

Column 1 lists the instantaneous value of voltage (or current). Thirty six values were computed (every five degrees of phase angle). There were totaled and divided by 36 to give us the average value of voltage or current. Average value is useful in electronic circuits. For example, the full wave rectifier where average electron drift is important.

Column 2 shows the rms value of the voltage or current in Figure 1, which is also the average value of the power in Figure 1. Each of the instantaneous values in column one were squared. The column was then totaled and divided by 36 (the number of instantaneous points computed). Computations were made to ten places with the pocket calculator. The average turned out to be precisely 0.5, the same value shown in the graphical construction above.

Since $W = I^2 R$ $W \max = I^2 \max (1) = 1$ watt peak

Since we are dealing with a one ohm load, each of the values in column two indicates instantaneous power as well as the square of the instantaneous current or voltage.

rms stands for root-mean-square. This means that we take the square root of the average of the squares. Since 0.5 is the average of column two, $\sqrt{0.5}$ or .707 equals the rms value of the voltages or currents squared shown in column two, and 0.5 represents the average of the power shown in column two.

It is important to note that each of the instantaneous values shown in column two represents the instantaneous current squared or the instantaneous voltage squared or the instantaneous power.

Some misinformed individuals have thought rms power to be .707 of the peak power. It sounds logical since rms voltage and current are .707 of their peak values. Let us examine the true value of rms power just to prove how untrue this really is. The figures in column two represent the instantaneous power in figure one. Column three shows the squares of each of the instantaneous power values of column two. The average of the sum of the powers squared is 0.375. The rms value of power is therefore, $\sqrt{.375} = 0.6123724357$. Quite a different value than .707, showing that rms power is not 0.707 times the peak power.

The 0.612 figure for rms power serves no useful purpose and I suspect has never been used for amplifier power ratings.

In dealing with power, we want to deal with the heating effect of a.c. as we deal with the heating effect of d.c. The watt is a unit of power (or rate of energy transfer), and is equal to:

10^7 ergs per second, or
3.4129 btu per hour, or
44.27 Ft-lbs. per minute, etc.

Therefore, amplifiers would be more properly rated for continuous sine wave average power (watts). rms (effective) values are fine for voltage and current but should not be used for power.

Recently, it has been implied that the use of the term rms power is just another ploy by sales departments to advertise inflated power ratings. If this were true, manufacturers would quote rms power figures as 0.612 of the peak power. However, many recent interviews reveal that this is not true. Manufacturers are measuring average power and then improperly advertising rms power. Since average power = 0.5 peak power and rms power = 0.61237 peak power (not .707).

$$\frac{.5}{.61237} = .816 \text{ to } 1 = \text{ratio of average to rms watts}$$

or

$$\frac{.61237}{.5} = 1.22 \text{ to } 1 = \text{ratio of rms to average watts}$$

Therefore, the manufacturer that measures 100 watts average power and quotes 100 watts rms power could legitimately quote 122 watts rms power. Current industry practice may be fallacious, but it is not dishonest.

1. The average value of a sine wave voltage or current equals .636 of the peak value.
   1a. The peak value of a sine wave voltage or current equals
   1.57 of the average value
2. The root mean square (rms) value of a sine wave voltage or current equals
   .707 of the peak value
   2a. The peak value of a sine wave voltage or current is equal to
   1.414 times the rms value
3. The square of the sine wave rms voltage divided by the circuit resistance equals the *average* watts (power), not rms watts

$$W_{avg} = \frac{E^2_{rms}}{R}$$

NOTE: This is the average, not the rms watts.

4. The square of the sine wave rms current multiplied by the circuit resistance equals the average watts (power), not rms watts.

$$W_{avg} = I^2_{rms} R$$

5. The product of the rms voltage and the rms current equals the average watts (power), not rms watts.

$$W_{avg} = E_{rms} \times I_{rms}$$

6. A wattmeter reads average, not rms watts.
7. The average power in a circuit is one half of the peak power.

$$W_{avg} = 0.5\ W\ peak$$

8. rms power is equal to .61237 of the peak value. There are no instruments (voltmeters, ammeters or wattmeters) that are calibrated in terms of rms power. Furthermore, it is a rather useless term.

$$W_{rms} = .61237\ W\ peak$$

9. Since
   $W_{avg} = .5\ W_{peak}$
   $W_{rms} = .61237\ W\ peak$
   .61237 = 1.22 to 1 = ratio of rms to average watts
   $\frac{.61237}{.5}$ = 1.22 to 1 = ratio of rms to average watts
   $\frac{.5}{.61237}$ = .816 to 1 = ratio of average to rms watts

TABLE I
VOLTAGE CURRENT AND POWER VALUES FOR FIGURE I
E max = 1 volt   I max = 1 volt   W max = 1 watt

Degrees	E inst. (volts) or I inst. (Amps)	I inst.² or E inst.² or W. inst. (Watts)	W inst.²
5	.0871557427	.00759612349	.00005770109
10	.1736481775	.0301536895	.00090924499
—	— — — —	— — — — —	— — — — —
175	.0871557427	.00759612349	.00005770109
180	–0–	–0–	–0–
TOTAL	22.90376554	18.	13.49999998
Average	$\frac{22.90376554}{36} = .636$	$\frac{18}{36} = 0.5$	$\frac{13.49999998}{36} = .375$
RMS	.636 to 1 = ratio of average to peak voltage or current	0.5 = .7071067812 .7071067812 to 1 = ratio of rms to peak voltage of current	.375 = .6123724357 .6123724 to 1 = ratio of rms to peak watts

the trials and tribulations of the novice kit-builder

Leonard Feldman

IN THE COURSE of testing and reviewing various high-fidelity products for AUDIO Magazine over the past several years I have, from time to time, had to review various electronic products which are offered to the consumer in kit form. I've watched as construction manuals of various manufacturers have become increasingly "easy to follow" and as recently as two years ago I would have stated positively that kit-building manuals had reached a level of perfection such that even an unschooled chimpanzee could, "in just two evenings of spare time," assemble a stereo receiver using only "simple tools such as a screwdriver, cutters, long-nose pliers and a low-wattage soldering iron." I would have taken this position, that is,—until I got involved in offering a kit product to the general public myself!

The kit product that I offered was, from my point of view, as uncomplicated as could be. There was only one printed circuit board to worry about, some 20 or so resistors and an equal number of capacitors, a few transistors and an IC, some coils, parts for a power supply, a single chassis plate for mounting everything, and a plastic cabinet. I wrote a detailed, 8-page construction manual, replete with "helpful hints," line drawings, check lists, step-by-step instructions and (for the more technical user) a schematic of the finished product. After all, I had the benefit of having read many "professional" kit-manuals in the course of my work and could borrow the tried-and-tested techniques of the best of them! I started to ship my kits with justifiable pride, offering to help "trouble shoot" any unsuccessful efforts for a flat $5.00 fee. Since then, I had the deflating opportunity to examine about 100 "unsuccessful efforts"—and I've rewritten the manual no less than three times! Now admittedly, in any true "cross section of the buying public" I knew I'd find a few weird interpretations—but some of the fiascos that were returned to us are worth describing, not so much for the amusement of the experienced kit-builder (though we'll admit that we very often were doubled up in laughter), but for the forewarning of the neophyte who is about to take "the plunge."

The Parts List versus Taking Your Chances

Most kit manuals start out by stressing the importance of opening up every package of parts and comparing the contents with the accompanying parts list. True, really elaborate kits usually subdivide the contents into individual envelopes or packages, but even the contents of one ot these individual packages can resemble the panic-producing melange shown in the photo of Fig. 1. If I were a kit builder embarking upon a first project, I'd rather skip the parts list check and take my chances on everything being there when I need it—still neatly packaged. I haven't run into a missing part for years, and if this misfortune should befall me in the future, I'll write to the manufacturer who would, I am sure, promptly make up the shortage. I find this approach much more practical than the piles of parts laid out on a workbench or table, especially since many electronic and hardware parts seem to be equipped with miniature motors which invariably propel them to the floor and don't shut off till the parts have rolled into a dark corner or under an immobile piece of furniture. The only caution I would offer is that you *not* throw away the shipping carton until all parts have been used up—as there may be one last envelope hidden amidst the packing material.

Timing Estimates

Some kit producers insist upon providing the kit builder with an estimate of the time required to put together their product. Obviously, the intent is to sell kits bv assuring the potential customer that he is not in for an endless task and I'm sure that in each such factory there is one (or more) technician who has the job of putting together a kit at breakneck speed so that the advertising agency can vociferously proclaim, ". . . build it in just one evening!" I've run into kits that would have to be transported to northern Greenland in the dead of winter to meet that requirement! Awake, kit manufacturers! DON'T start out by intimidating your potential customers. They're going to try to equal or beat your time estimate and in doing so, you'll find yourself with more kits improperly wired or incomplete and your trouble-shooting staff will soon be larger than your instruction manual writing staff! Awake, kit builder! The finished product you're trying to complete will probably give you good service for years to come—don't push it!

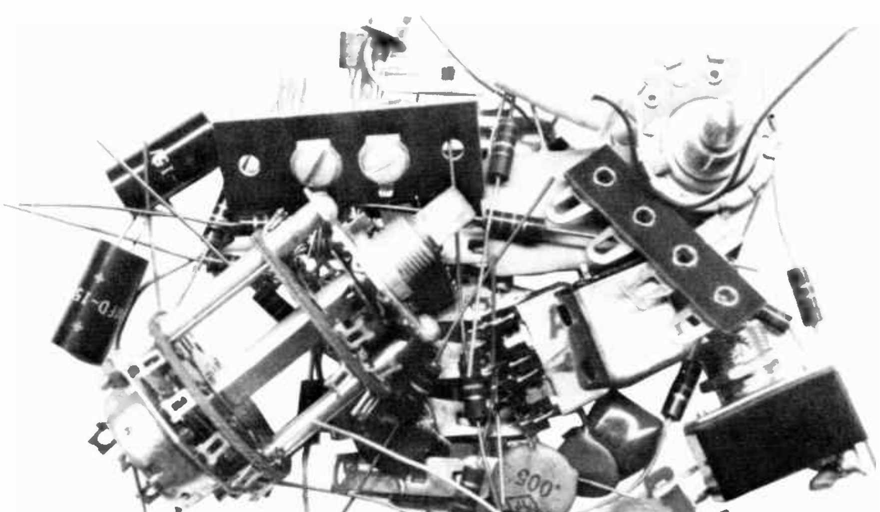


Fig. 1—HELP! Where do I begin?

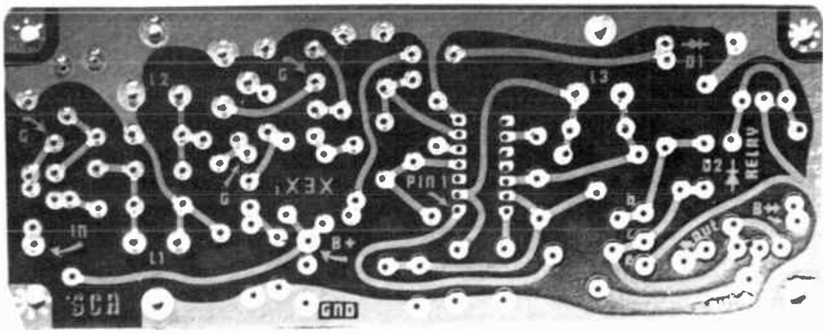


Fig. 2—The soldering side of a p.c. board.

"Only a Few Simple Household Tools . . ."

Many kit manuals approach the subject of a soldering iron apologetically. The presumption, I suppose, is that most "households" are not normally equipped with a proper soldering iron for kit building. They therefore try to imply that a single 35 or 45 watt iron is all you need. Some even imply that a "soldering gun" can successfully be used for the multiple kinds of soldering required to put most kits together. Nothing could be further from the truth. The very way in which a "gun" works will *destroy* the fragile copper patterns of a printed circuit board. Proper soldering of components on a printed circuit board involves the speedy but concentrated application of heat to the junction between the copper pattern and the component lead end emerging from the printed circuit board hole. A gun cools off quickly and reheats so slowly when the trigger is pulled that you end up holding the oversize "business end" of the gun against the printed circuit board for an agonizing length of time—often just long enough to separate the copper pattern from the base material of the board itself. We have found that a single, pencil-tip iron *can* do the entire soldering job for kits, *providing* you have more than one tip for it. Soldering of p.c. boards should be done using a pointed tip which flares to a diameter no greater than ⅛ inch. Heavier, chisel-shaped tips should be used for soldering directly-wired connections, such as wires to terminal strips, wires to socket terminals, etc. If there is any "chassis grounding" to be done (soldering of electrolytic capacitor mounting tabs to chassis, etc.), an even larger tip may be required and, in some cases, a higher wattage iron may be required (65 to 90 watts).

Before we leave the subject of soldering as related to printed circuits, we must tell you about one of the kits that was returned to us about a year ago. Upon opening the carefully packed unit, we discovered, to our utter shock and disbelief, that the novice kit builder had inserted *every single part* in the p.c. board from the *wrong side*. That is, resistors, capacitors, transistors and IC's were all sitting nicely on the *copper pattern* side of the board. With all due respect to this kit-builder, we must say in his defense that he had somehow managed to *solder* every part from that side, by leaving a bit of exposed lead above each hole entry point. Needless to say, the board had to be scrapped. We offered to replace the board and all associated parts at our cost, but I suspect the customer was too embarassed to accept our offer and may, to this day, be carefully unsoldering each part and installing it from the correct side. The story may sound incredible, but when you stop to think about it, some manufacturers show board layouts as viewed from the copper side of the board, while others show layouts from the component side, with an X-ray view of the copper below, for purposes of "clarification." Make sure you understand the presentation in your kit manual before you start or it could happen to you. The really funny thing about this catastrophe is the fact that if we were dealing with passive parts alone (resistors, capacitors, coils, etc.), the board might actually have worked, but since there were three-lead transistors and an IC involved, "bases" quickly became "collectors" in the mirror-image situation that resulted. Without belaboring the point, Figs. 2 and 3 show the P.C. board involved and parts are normally inserted with the Fig. 3 view facing you.

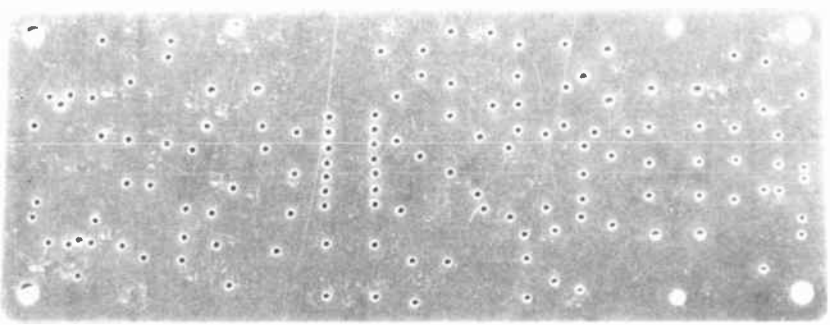


Fig. 3—The parts-insertion side of the same p.c. board.

The list of tools usually recommended besides the controversial soldering iron consists of a screwdriver, cutting pliers, needle nose pliers (sometimes called "long nosed pliers"), and a ruler (for measuring prescribed wire lengths). It usually comes as somewhat of a shock to discover that most household screwdrivers have blades which are far too thick and broad to get into the slots of the miniature screws which usually accompany most kits. Often, Phillips-head screws are supplied, and these usually stop the kit-builder dead in his tracks. I have found myself using no fewer than four different sizes and types of screwdrivers in building the average kit and having a variety of types and sizes on hand is a great convenience. I find, too, that the tool which I use more than any single screwdriver is one which is hardly ever mentioned—the "nut driver" (also popularly known by its trade name of Spin-tite). This handy gadget (shown in Fig. 4) does for a nut what a screwdriver does for a screw. In mounting parts using machine screws and nuts, it is far easier to hold the screw stationary (with the blade of the screwdriver) while turning the nut with the nut driver than to rotate the screw with the screwdriver while trying to hold the nut stationary with oversize finger tips that never can get into the tight places required. A ¼ inch and a 5/16 inch nut driver will cover most requirements, both dimensions referring to the distance between the parallel "flat" sides of the nut in question.

Although much of the work of kit building has been reduced to insertion of parts into p.c. boards, there still remain a goodly number of loose wires that are used to interconnect from one p.c. board to another or for interwiring of such sections as power supply, power output stages in an audio amplifier, and other sections which do not

Fig. 4—A nutdriver used *with* a screwdriver helps assembly of hard-to-get-at screws and nuts.

lend themselves readily to p.c. board layout. What kit builder has not dreaded the familiar phrase, "strip ⅜ in. insulation from each end of a 12 in. piece of wire"? I must confess that I have been "stripping the ends" of wires with cutting pliers for nigh on to 20 years and every now and again, the wire in question becomes, successively, 11⅞ in., 11¼ in., 10⅞ in., 10½ in., etc. as the sharp tips of the cutting pliers bite into the wire itself instead of just "stripping the insulation." A $1.98 solution to this problem is shown in Fig. 5. It's called (as you might have guessed) a wire stripper, and while there are varieties of this product that sell for several dollars, the type shown not only strips all sizes of wire you are likely to use (it's adjustable), but it doubles as a wire cutter too. About the only time I find I need cutting pliers now is for trimming the component leads of parts I have inserted into p.c. boards, where the pointed

Fig. 5—A handy tool for stripping and cutting hook-up wire and reducing kit-building frustrations.

tip of the cutting pliers lets me get in close enough to cut off excess lead lengths so they don't "spill over" to adjacent copper patterns.

"Never Skip Ahead Of The Instructions . . ."

Here, at last, is the one admonition given by kit-manual writers with which I must fully concur. I must confess that I too have been guilty of the cardinal sin of kit building—jumping ahead of the given order of step-by-step instructions. Results have varied from the relatively mild punishment of having to unsolder several well-soldered connections (to insert additional parts that were destined to go to the same terminal) all the way to having to unwire and remove a 12-lead power transformer because I just couldn't squeeze a fuseholder past its bulky mass to mount it on the back chassis flange. Let's face it, friends, despite fragmentary evidence to the contrary, most kit products have actually been assembled by their designers at some time or other and the printed assembly and wiring manuals are an outgrowth of that experience.

"Now for Alignment and Test . . ."

There is nothing quite so sad as completing a high fidelity component kit, checking all connections, turning it on, obtaining what seems like satisfactory performance and then proceeding to louse things up by second-guessing the alignment and test procedure. (Actually, I suppose an even sadder result is turning it on and having *nothing* happen at all. No, in the last analysis, I guess the most tragic event is turning it on and having something happen that isn't supposed to happen—like smoke pouring forth from the innards, or fuses blowing—or—well, let's not even pursue this any further . . . .)

"It Doesn't *Look* Factory Wired . . ."

Unless you want to invest thousands of dollars in automatic wave-soldering units, automatic terminal connecting machines, riveting and eyeleting machines, solderless terminal connecting machines, tension wire-wrap machines, and the like, don't be surprised if your "end product" still looks a bit crude compared to the "factory wired" job. The somewhat archaic methods of wiring and assembling detailed in kit-manuals persist not because kit-designers are oblivious to manufacturing progress, but rather in recognition of the limitations imposed by the needs of the "home builder" who is not expected to "capitalize" his kit building endeavors to the degree that a manufacturer might.

Happily, most kits today can be aligned with *no* test equipment at all. If that's the case, go to it, treating each successive step as if it were still part of the assembly and wiring process. If a voltmeter or ohmmeter *is* required, don't proceed without one. If 'scopes and generators are required, don't try to bypass their use—even if it means taking your finished masterpiece to a friend who has such equipment. To salve your "pride of having done it yourself" (one of the main selling features of kits), perhaps he's a good enough friend to let *you* do the test and alignment, albeit with *his* equipment. If he insists on being the "engineer," that's still better than the thought of destroying your "one evening's work" (or one week's work, or one month's work) over a few minutes of checking and testing.

At the risk of further intimidating the faint-hearted (an outcome hardly initially planned for this otherwise helpful treatise), I'd like to conclude by quoting a paragraph from the section of my most recently acquired kit-manual entitled:

"In Case of Difficulty"

After dealing with a host of trouble-shooting hints, measurements that can be made and areas that can be visually checked and re-checked, the section concludes (as do I) with this immortal thought:

NOTE: In an extreme case, where you are unable to resolve a difficulty, refer to the service and warranty section of the "kit builders guide" and to the "factory repair service" information on page 143 of this manual.

Equalization of Sound Reinforcement Systems

Daniel Queen

RECENT INTEREST in equalization of sound reinforcement systems has resulted in many terms being tossed about. Included have been broadband equalization, narrow-band equalization, notch-filters, one-third octaves, direct-to-reverberant ratio, etc.

There is much dispute now on the relative merits and effects of these various measures. Before a sound contractor decides on a particular measure, he should know exactly what that measure is capable of doing but more importantly, what it is *not* capable of doing.

The technology of using notch filters for room equalization was popularized by C. Paul Boner after studying church reinforcement systems which appeared to exhibit several sharp response modes. When these modes were removed by means of filters only a few Hertz wide, the capability of the system to provide gain before howlback was enhanced. Boner's studies showed that many of the rooms with which he worked could be improved using as many as fifty or sixty such filters. It should be noted that Boner advocated the use of such filters only after the system had been built with high quality components and after the system was broadband equalized.

Herein lies the rub: The difference between broadband equalization and notch filtering is sometimes confused. Broadband equalization is accomplished after making measurements of the room response, usually by means of third-octave filters, then correcting the response in each of these third-octave bands. The usual approach is to make the third-octave measurements at various points in the room. A noise source feeds the amplifier or microphone. The response at the various locations is averaged, then estimates are made about which are the most significant factors to be corrected.

The equalization filters used to correct the system in this process are no narrower than one-third octave. In the vicinity of 1 kHz such a filter is 200 Hz wide. In contrast, narrow-band equalization is achieved by increasing the gain of the sound reinforcement system until either ringing or howlback occurs. A notch filter is then inserted at the frequency of the ringing or howlback. Such a filter is typically about 5 Hz wide at its half-power points.

Sources of Perturbations

Deviations from flatness in a sound reinforcement system, when in an actual room, are caused by characteristics of the room itself, of the electroacoustic components, and of the interaction between the room and the electro-acoustic components.

Every room acts somewhat like an organ pipe. For each major dimension, resonance frequencies occur at the frequency where the dimension is one-half wave length and at integral multiples thereof. Thus, a room 30 feet in length will have resonances at 15 Hz, 30 Hz, 45 Hz, etc. The other major dimensions will generate resonances at similiar harmonic intervals. One can see that, as frequency increases, more of these resonance modes will occur (the "mode-density" will increase). In fact, in most rooms, the mode-density above 250 Hz is such that modes are separated by no more than 5 Hz, their Q is approximately equal, and they are distributed evenly about the room, resulting in an essential flat amplitude vs. frequency response when exited. Only when rooms are unusually shaped or have a very uneven distribution of absorbing materials, can one expect the amplitude of any mode to exceed the average of all the modes—particularly within third-octave bands.

Thus, except at very low frequencies, the room itself contributes little to nonflatness due to individual modes. Instead the room characteristically shows greater absorption and therefore amplitude roll-off at the extreme ends of the band. The roll-off at the low end is mainly due to vibration of walls, causing sound to leave the room, and at the high end by the absorption of materials on the walls or in the room. The effect of such absorption is usually spread over a wide band, looking much like the effect of tone controls (Fig. 1).

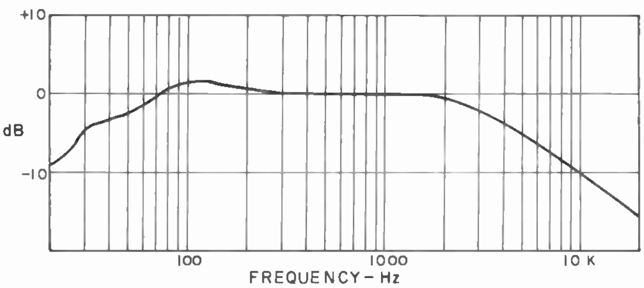


Fig. 1—The broadband steady-state response of a typical meeting hall (averaged over tenth octaves).

The equalization for it is broadband. However, an unusual effect is caused when a loudspeaker or a microphone is inserted into a room. If the loudspeaker is spaced away from a reflecting surface, it will produce, a series of peaks and dips associated with the distance from the surface, as shown in Fig. 2. These peaks and dips go to a minimum when the speaker is actually put into the wall. Usually these peaks and dips are well inside the pass band of a third-octave filter and must be equalized by means of narrow-band filters.

A similar effect occurs with a microphone, the nearest reflecting surface being the floor or a lectern table top. Similarly the effect is minimized by moving the microphone close to or into the table top or if possible, close to the floor. The effect of floor reflections can be minimized by use of directional microphones provided they are truly directional in all planes (as we shall discuss later.)

Amplifiers and Signal Processing Circuitry

The amplifiers and signal processing circuitry in a sound reinforcement system generally can be depended upon to be free of amplitude perturbations. However, care must be taken that, due to unusual tone control circuitry, etc., excessive phase shifts that could throw a system into positive feedback do not occur.

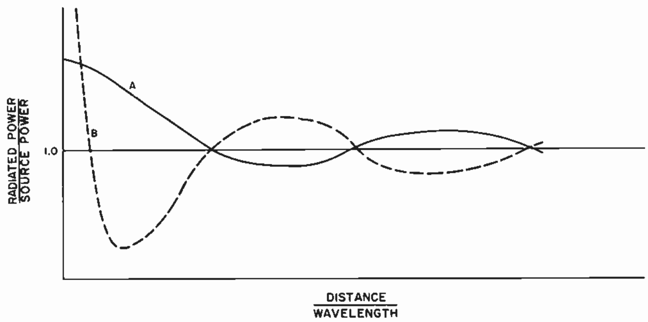


Fig. 2—The effect of locating a loudspeaker near a reflecting surface. Curve A is for a hard surface; curve B for an absorbent surface. For any fixed distance, the curves represent the amplitude-frequency characteristic of the radiation. The sensitivity-frequency characteristic of a microphone follows similar curves.

Loudspeakers and Microphones

The characteristics most commonly specified for microphones and loudspeakers for sound reinforcement systems are frequency response and directivity. These should be the characteristics which would most affect the susceptibility of the system to howlback. The freer of peaks the response is, the less the chance of howlback at one of those peaks. The more directive the transducers can be, then the more direct sound from the talker and to the listener will be produced at a given gain setting of the system. Thus it would seem that a microphone specified with a flat response and with good directivity would be the best microphone to choose for the system. Unfortunately, the frequency response given with the microphone is usually the zero incidence response in a free field, that is, a non-reflective environment. If it is a directional microphone, the response at one or two other angles may be plotted. Loudspeakers are specified with the on-axis response supplemented by a polar plot of the loudspeaker at 4 or 5 frequencies. The polar plots are usually shown in two planes around the loudspeaker. This would be adequate to tell the performance if actual rooms had only the two planes measured in these response plots. Actual rooms, however, must be analyzed in three dimensions, so no single incidence or plot in a few discrete planes reveals the full performance of the transducer.

Incident and Reflected Sound

The problem can be better appreciated through an examination of the different sound fields that exist in an actual room. The fields are defined by time and direction: 1. The sound that goes directly from the sound source to the receiver; 2. The sound that goes from the sound source to the receiver with one or two reflections, and 3. The sound that goes from the sound source to the receiver with an almost infinite number of reflections. These are called, respectively, the direct field, the early reflections, and the reverberant field (Fig. 3). In most of the literature the early reflections and the reverberant field are differentiated on the basis of the sensitivity of human hearing to the relative time of arrival of sounds. If the reflected sound is heard less than 65 milliseconds after the direct sound, the human hearing process fuses it with the direct sound. After 65 milliseconds the human hearing process hears it as an echo or a discrete sound.

Speech intelligibility is usually found in the direct sound and the early reflections. Recent studies have shown that a low ratio of direct-early to reverberant sound acts similarly to a low signal-to-noise ratio in reducing intelligibility. Contrariwise, musical quality is found in all fields. Thus a non-reverberant room, which sounds very good for speech, would sound dead for music.

One of the reasons for this lies in the transient characteristics of a room. The direct sound flows from the sound source to the receiver at the propagation velocity of sound in air, while dropping in amplitude 6 dB for each doubling of distance. The early reflections travel similarly, although the distance, naturally, is greater. However, the reverberant field acts like a capacitor being charged. In fact, its rate of build up is very similar to the exponential charging rate of a capacitor.

Thus the rapid transient sounds of speech will not excite the reverberant field as much as the continuous sounds of music. A room, which in combination with a loudspeaker, is found to have a direct-early to reverberant energy ratio at the center seats of 0 dB, when measured by steady state methods, may actually have a 10 dB direct-early to reverberant energy ratio for speech.

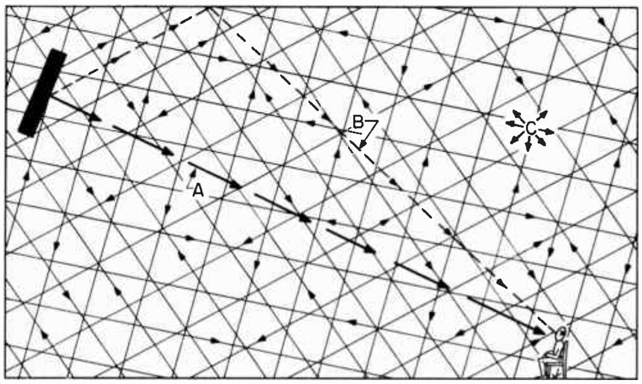


Fig. 3—The sound fields excited in a room by a loudspeaker. A is the direct sound; B, the early reflections, and C, the reverberant field.

Therefore, for speech intelligibility, one would be most concerned with the response of a loudspeaker and microphone in the direction that would be normally between the sound source and the receiver. This would seem in concurrence with the specification of the microphones and loudspeakers in terms of frequency response at zero incidence.

The Acoustic Feedback Loop

However, good response in the direct field is meaningless if one cannot turn the gain up without getting howlback. Howlback occurs because, in the feedback loop shown in Fig. 4, the gain between the output of the loudspeaker and the input of the microphone becomes greater than one and in-phase at a particular frequency. If the phase and amplitude characteristics show some non-linearities, which they usually do, the system can ring without howling. It is obvious that this howlback causing path is not in the on-axis direction, since a microphone is usually not placed directly in front of and facing the loudspeaker.

Direct-To-Reverberant Ratio

As shown in Fig. 3, the amplitude in the direct field drops in half with every doubling of distance. In contrast, when the sound source is continuous, that is steady state, the amplitude of the reverberant field is almost the same throughout the room (although peculiar shapes and unusual reflecting

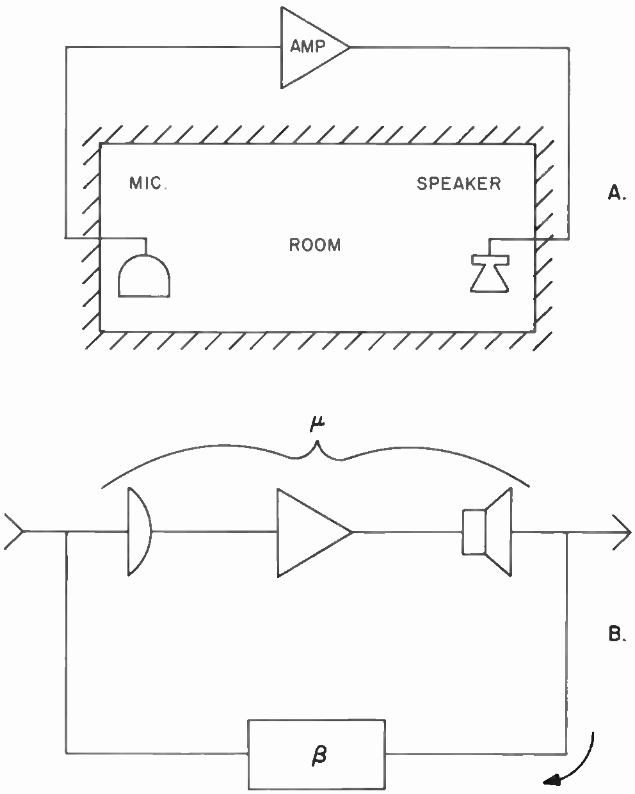


Fig. 4—Sound reinforcement system, A, and its electrical analog, B.

There are two ways to measure the response of a microphone. One is the free-field response, in which a source closely simulating a point-source is placed at a fixed angle to the microphone in a room free of reflections at a distance many times the largest dimension of the microphone. Using this method, the actual frequency-response is measured at various angles to the microphone. Thus its directivity pattern can be plotted.

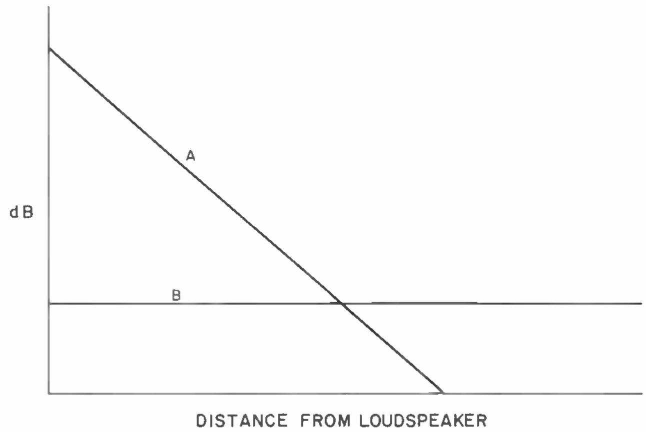


Fig. 5—The relation of sound fields to distance from the loudspeaker; A, the 6 dB loss per doubling of distance for the direct sound, and B, the reverberant field.

or absorbing surfaces can cause some variations). Thus, as one moves away from the sound source the amplitude of the direct field approaches that of the reverberant field (Fig. 5).

Where the microphone and loudspeaker are completely directional, so that the loudspeaker radiates only in a forward direction and the microphone gathers only from its front, then, if the microphone and loudspeaker are not facing each other, the only path between the microphone and loudspeaker would be the reverberant field. It would be only the peaks in the reverberant response of the room that would cause howlback problems (provided we selected microphones and loudspeakers with flat response). Yet, we have shown that above about 250 Hz the room should be essentially flat (except for broadband variations). Why then is it still found necessary to use narrow-band filters? The answer can be seen by examination of the off-axis characteristics of the transducers.

Random Incidence Versus Axial Response of Microphones

The directional patterns are commonly designed into microphones as shown in Fig. 6. Although the figure shows them in two dimensions, they actually exist in three dimensions. The omnidirectional pattern is ideally a sphere with the microphone at its center; the cardioid pattern roughly the shape of a cherry with the microphone at the stem; the bi-directional pattern, two spheres end to end with the microphone between them; and the ultra cardioid pattern roughly the shape of a cucumber with the stem projecting behind the diaphragm. Ideally, these patterns are the same at all frequencies. However, just as there are no point sources in the real world, there are no ideal patterns in actual microphones.

The second method is the random-incidence (sometimes called pressure) response. In this method the microphone is placed in a highly reverberant room at a point where it will receive very little direct sound from a source but will receive an infinite number of waves from an infinite number of angles to it. Using the randon-incidence method, all the free-field responses from every angle are averaged. The difference in sensitivity level of a microphone between perpendicular incidence and random incidence is a measure of the perpendicular incidence directivity-factor of that microphone. It is obvious that a true omnidirectional microphone would have a directivity factor of 1, whereas the directivity factor of an ultra-cardioid microphone would be higher than that of the other types. However, for a microphone to respond to vibrations in the air, it must have a large enough diaphragm to absorb sufficient energy to create an electrical signal sufficiently above the thermal noise of the microphone structure to result in a useful signal-to-noise ratio.

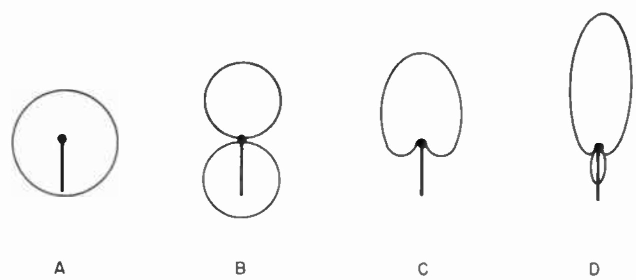


Fig. 6—Ideal microphone directivity patterns: A, nondirectional; B, bidirectional; C, cardioid, and D, super-cardioid.

Therefore, most omnidirectional microphones have diaphragms with diameters in the order of an inch. Figure 7 shows that when a wave having a length close to that of the diameter of the diaphragm falls on the front of the diaphragm it will respond the same as a wave having a length much larger than the diaphragm. However, if the wave comes in parallel

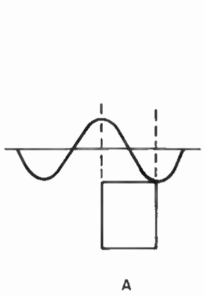


Fig. 7—A, a wave striking a microphone diaphragm with parallel (90°) incidence; B, a wave striking the same diaphragm with perpendicular (0°) incidence. In case A, simplified for illustration, the diaphragm sees equal negative and positive pressure, causing a null. However, in case B, the perpendicular incidence, the diaphragm sees only the wavefront, so no null occurs at any wavelength.

to the diaphragm, it will tend to cancel itself out as it passes over the diaphragm. Thus, a microphone which is designed to be flat on-axis actually must have a hole in its response off-axis. If, on the other hand, it is designed to be flat in its random-incidence response, it will have a peak somewhere around 5 kHz on-axis. In one case, the microphone can be seen to have a non-flat response in the reverberant field, aggravating the howlback; in the other, the direct-early sound with be peaked and raspy.

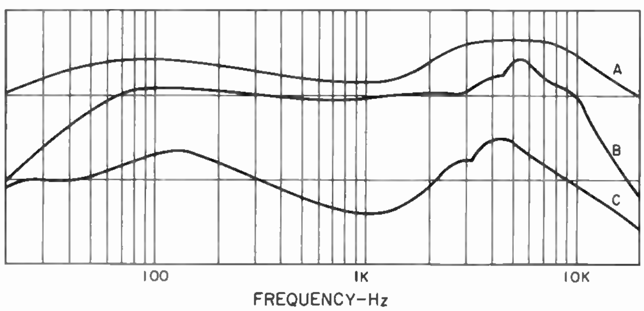


Fig. 8—Sensitivity-frequency characteristics of a cardioid microphone. A, random noise incidence; B, on-axis (0°) incidence, and C, rear axis (180°) incidence.

Figure 8 shows the response of a typical, high-quality cardioid microphone. The on-axis response is shown to be very flat. The off-axis response at 180° shows some variations including peaks near 200 Hz. It is this off-axis characteristic which distorts the response in the reverberant field. It is possible, in fact, that the peak in the off-axis response of a directional microphone can be sufficient to put the loudspeaker into the direct field of the microphone, that is at that frequency, the signal directly from the loudspeaker can be louder than the signal that flows to the microphone from the loudspeaker through the reverberant field.

Loudspeakers

The directivity patterns of loudspeakers are even more subject to this type of inconsistency in directional pattern. Because loudspeakers need radiating surfaces which are large

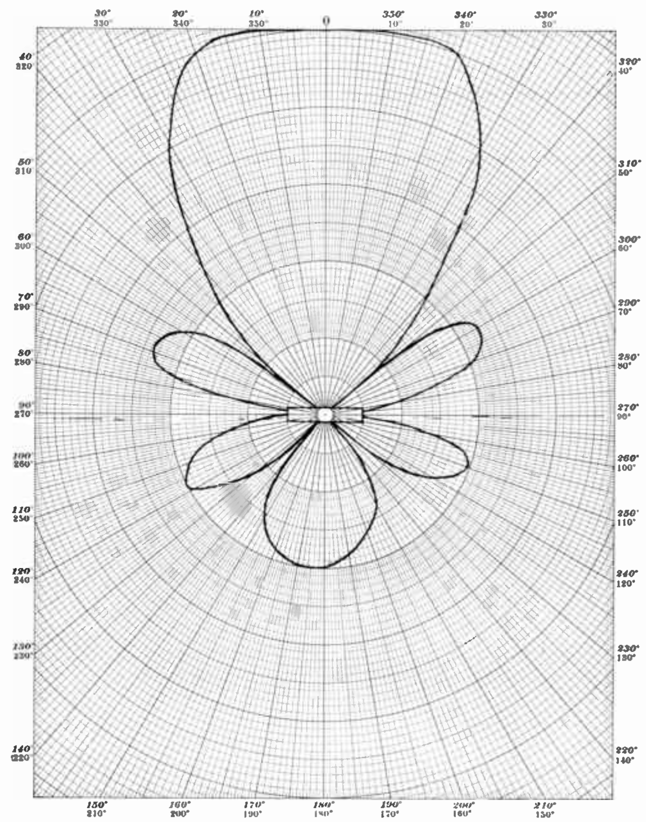


Fig. 9—Polar plot of loudspeaker radiation at one frequency on one plane.

compared to the shorter wave-lengths they must radiate, their radiation can be made consistent only by approximate methods. Thus every loudspeaker, even if its individual elements are nearly ideal, has many finite lobes as shown in Fig. 9.

Equalization for Defects

The peaks, which cause howlback at a lower gain than could be accomplished with ideal transducers, can be reduced with narrow band filters—but care must be used when doing so. Figure 10 shows the effect of a 10 Hz filter on a peak 30 Hz wide. Since the filter is placed in the signal processing circuitry of the sound system it also affects the direct sound that is delivered to the listener. The audible effect of the 10 Hz filter is very small and would probably not change the sound heard by the listener. Figure 11 shows the result of adding additional filters to reduce the peak even more.

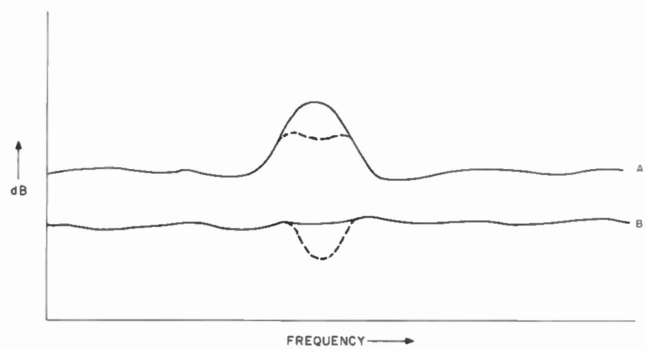


Fig. 10—Effect of a 10 Hz wide notch filter on A, reverberant field, and B, direct-early sound field.

As these additional filters are added, a hole begins to develop in the direct field response causing deterioration in the quality of the sound at the listener. It is for this reason, among others, that acousticians strongly warn against beginning narrow band equalization until one has carefully chosen and carefully installed the best transducers.

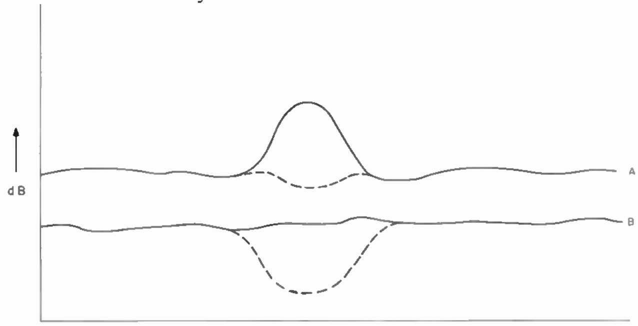


Fig. 11—Effect of additional filters on A, reverberant field, and B, direct-early sound field.

Directivity Pattern Defects

As with microphone response, the response of loudspeakers can be measured in two ways. A test microphone may read the response from different angles to the loudspeaker in a free field. Similarly, the loudspeaker may be used to excite a reverberant room with a microphone placed so it picks up the reverberant field without receiving the direct field of the loudspeaker. The first method measures the free field response of the loudspeaker. The second measures the reverberant or power response of the loudspeaker. The power response is the characteristic that excites the reverberant field in an actual hall. Frequently, due to the lobes in the loudspeaker polar response or due to problems in the placement of the loudspeaker in the room, the power response of the speaker is very different from the direct field response. If, when measuring the response of a sound system preparatory to broadband equalization, the microphone is placed in the reverberant field of the loudspeaker and equalization is applied, the direct field of the loudspeaker may be severely distorted affecting the quality of the sound. (Fig. 12).

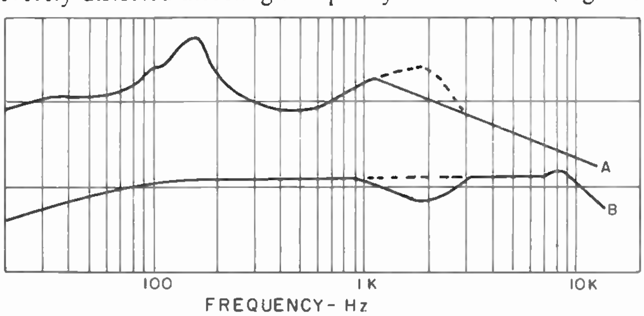


Fig. 12—Possible effect of equalization of the direct-early sound field (B) on the reverberant field (A).

If, on the same loudspeaker, the microphone is placed close to the loudspeaker, in the direct field, then the reverberant field may be distorted to such a degree as to aggravate the howlback of the system (Fig. 13).

Effectiveness of Equalization

Thus, it can be seen that a system with severe transducer problems, that is, a substantial difference between the free field and reverberant field responses of the transducers cannot successfully be equalized unless that equalization can be applied separately to the reverberant and direct fields—a condition which is rarely practical and is difficult to achieve.

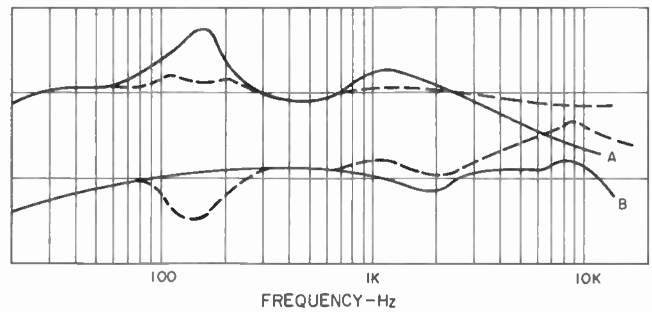


Fig. 13—Possible effect of equalization of the reverberant field (A) on the direct-early sound field (B).

To produce a system capable of high gain before howlback, the sound system designer must first obtain representative off-axis response curves for his transducers and compare these with the reverberant field and with the on-axis response. He must be sure that the combination of microphone and loudspeakers he chooses will not cause severe peaking in the reverberant field.

Secondly, the designer must take care in the placement of his microphones and loudspeakers so that he does not cause peaking due to reflective surfaces. He may then proceed to broadband equalization.

Broadband equalization may be achieved by placing the system microphone in the direct field of the loudspeaker, that is, close enough to the loudspeaker so that it will be mainly direct sound that is measured. Third-octave or broader filters are then inserted to flatten the response curve. Note that because the high frequency response in the reverberant field usually falls off as in Fig. 1, flattening the response with a microphone in this field will cause high frequency emphasis in the direct field resulting in a raspy sibilant sound.

Having thus flattened the direct field of the sound system, the system gain should be increased until howlback occurs. Narrow-band notch-filters are then inserted to reduce the susceptibility to howlback. Such filters should be inserted until they begin to reoccur around the same frequencies. At this point the designer should go back to the direct field and see that his broadband equalization has not been distorted by the insertion of excessive narrow-band filters.

Conclusion

Sound system equalization should only be used as a last step in the design of a sound system and should never be used when care has not been taken in the selection and placement of loudspeakers and microphones. For speech reinforcement an effort should be made to place the direct sound of the talker at the ears of the listener. To accomplish this, microphones and loudspeakers should be as directional as possible, provided their off-axis response is as uniform as their on-axis response. If this is achieved, the improvement in gain before howlback achievable with directional loudspeakers and microphones will be proportional to the directivity factors of the loudspeakers and microphones. Improvements of gain in excess of 20 dB over systems commonly in use can be achieved by proper selection and placement of loudspeakers and microphones. The next step, the process of equalization, can achieve additional gains in the order of 6 to 15 dB.

References

1. C. P. Boner and C. R. Boner, "A Procedure for Controlling Room-Ring Modes and Feedback Modes in Sound Systems with Narrow-Band Filters," J. Audio Eng. Soc., Vol. 13, No. 4 (1965).

2. C. P. Boner and R. E. Boner, "The Gain of a Sound System," J. Audio Eng. Soc., Vol. 17, No. 2 (1969).

Equalization in the Home

John Eargle

THE MOST significant advances in home music listening in the last five years have been the advent of quadraphonic sound, the general adoption of B-type Dolby noise reduction, and the "normalization" of loudspeaker/room response through system equalization. While quadraphonics represents a distinct *revolution* in the listening experience, noise reduction and system equalization are *evolutionary*, providing only improvements in what we may call the "transfer function" from studio to listener.

Our increased concern with these improvements stems mainly from the revolution which has been going on in the area of musical values; today, about 90 per cent of recorded product in this country is rock/pop, and virtually *all* of this product has its genesis over loudspeakers in the control room. Thus, noise reduction and equalization can justifiably be sought as ends in their own right.

Quadraphonic sound is happening at every turn, and the 15-year-old mono-stereo controversy is being re-enacted all over again. Noise reduction is being introduced gradually, and in a comparatively orderly fashion. It has long been an important part of the original recording process. A while ago it found its way to the cassette medium—one which was sorely in need of it. It has recently found its way into the production of reel-to-reel tapes, and it can make this medium, in either a stereo or quadraphonic configuration, the superlative one it was meant to be. It has also been introduced into FM broadcasting, and there it promises greater effective coverage for classical music stations.

The evolution of home sound system equalization has been neither orderly nor clear-cut in its direction. First of all, it has been, and will likely continue to remain, a fairly expensive "embellishment" on what is already a good sound system. Many systems, in many good listening rooms, simply do not need specific contouring of the system to the room. On the other hand, a marginal low-powered system *cannot* be equalized effectively at all.

A problem which has plagued the general adoption of home equalization has been one of adequate instrumentation—and who is to man that instrumentation. Sound level meters are rare; ⅓-octave noise tapes are rarer—and Real-Time Spectrum Analyzers are not only rare, but expensive as well. It seems that the average audiophile is expected to equalize his own system *by ear*, since the number of dealers willing to provide the service is suprisingly low.

Let us now consider in detail the particular goals of equalization, some of the specific hardware available for the purpose, and instrumentation necessary to equalize accurately.

The Goal Of Sound System Equalization

Most professionals consider sound system equalization to be the answer for the audiophile who wants to duplicate in his listening room the acoustics and ambience of the recording studio control room. This is an ambitious goal, and it can, quite honestly, be met only through the use of detailed ⅓-octave equalization. Before this can be done, care must have been taken that the power-handling capabilities of the speakers and amplifier at hand are equal to the demand; both the watts and the ability to *handle* the watts must be duly assessed. It's quite an investment—but one well worth it for the dedicated audiophile.

A second function of equalizers is met by the myriad one-octave devices which are available. These operate in a sense as "super" tone controls—such devices have perhaps nine to

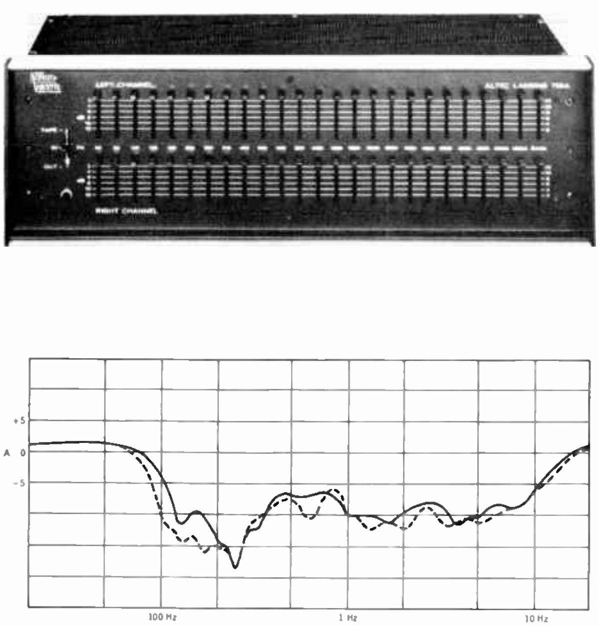


Fig. 1—The Altec Acousta-Voicette one-third octave room response equalizer and two typical electrical equalization curves produced by it.

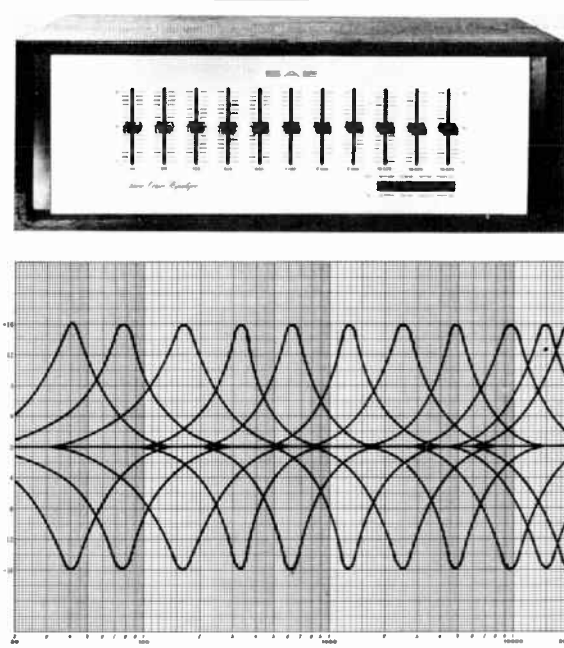


Fig. 2—The SAE MARK VII Stereo Equalizer and its family of curves.

eleven vertical slide controls which allow the user to tailor his system by ear. They are indeed useful devices, considering the variety of monitoring conditions between studios and the resultant spectral variations in records.

A typical example of a ⅓-octave equalizing system is shown in Fig. 1. This device, the Altec Model 729 Acousta-Voicette, enables the acoustical contour to be continuously and smoothly manipulated between the frequency extremes of 63 Hz and 12.5 kHz. It is a stereo unit and is normally inserted at the tape-out/monitor-in terminals of a typical stereo preamplifier or receiver. The Model 729 has the normal tape-in/tape-out monitoring facility which provide total flexibility in interfacing this device with a high-quality stereo system.

Typical of a one-octave wide equalizer is the SAE Mark VII Octave Equalizer. This device is available in a stereo model capable of handling the frequency range from 40 Hz to 20 kHz. Another octave-wide device is the Shure Model 610 Equalizer. Whereas the SAE device is a *graphic equalizer*, offering peaking as well as dipping functions for each vertical slide control, the Shure device, like the Altec 729 offers a family of combining-type attenuation-only curves. Figure 2 shows the SAE Mark VII along with a family of typical response curves for the device.

Instrumentation

Fig. 3—The 8050A Real Time Analyzer, a device which provides continuous monitoring of sound energy levels in one-third octave bands. The response, which is updated each 30 milliseconds, can be viewed in either *fast* or *slow* modes.

Real time analysis has greatly simplified the equalization process in both the consumer and professional areas. The Altec Model 8050A Real Time Analyzer is shown in Fig. 3. The device, introduced in 1971, has reduced the cost of real-time analysis to almost one-third of what it had formerly been, and this means that hi-fi dealers as well as professional sound contractors can perform detailed equalization at moderate cost to the consumer as well as moderate investment for themselves. Figure 4 shows the response curves of the individual filter sections which are displayed on the screen of the 8050A.

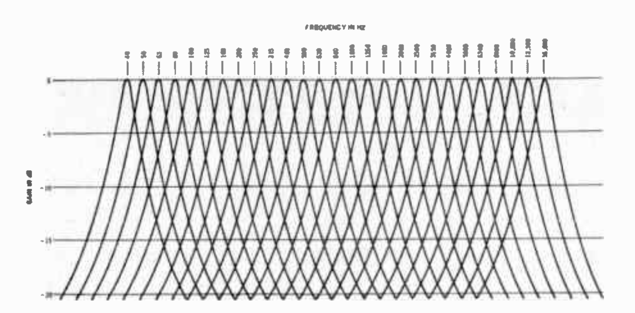


Fig. 4—Filter response curves for the 8050A Analyzer.

The Interaction Between The Loudspeaker and Listening Room

A typical high-quality loudspeaker system located in a 20' x 15' x 9' living room may present a pre-equalization response as shown in Fig. 5-A. A pink-noise generator, which provides a random noise signal exhibiting equal energy-per-octave, is applied to the monitor input of the system.

One may be surprised to see such a wide range of response —on the order of 16 dB—but this is typical of many high-quality systems in many rooms. The inverse curve was introduced with the Altec 729 equalizer flat out to 8 kHz, with a 6 dB/octave roll-off above that point. Some kind of controlled high frequency roll-off is desirable to compensate, at least in part, for the fact that most records are themselves mixed and mastered over systems exhibiting a degree of high frequency roll-off. This is a standard situation throughout the industry; what is *not* standard is the precise degree of roll-off.

The 16 dB response variation, once it has been corrected, calls for a 40-to-1 power ratio, and consequent power reserve, between the equalization extremes. Specifically, in order for the left channel of the system of Fig. 5 to produce the same response level at 40 Hz as it does at 250 Hz, it will require *40-times* more power; if 2.5 watts are sufficient at 250 Hz, then 100 will·be required at 40 Hz. It has long been observed that the prime problem areas in most equalized systems are the insufficient provision of low frequency power as well as the inability of the loudspeaker to handle such powers. As we have said earlier, only *good* systems can be equalized.

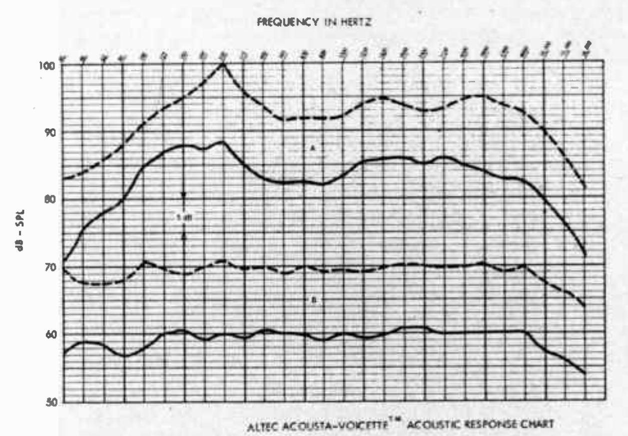


Fig. 5—Typical before and after acoustical response curves for a stereo system in a normal environment. Equalization provided by an Altec 729 Acousta-Voicette monitored by an 8050A Analyzer.

"Pre-Equalized" Loudspeaker Systems

More and more, we are seeing loudspeakers in the marketplace with their own complementary active equalizers. These devices, placed just ahead of the power amplifier, provide fairly broad correction for the most commonly met loudspeaker/room difficulties. The usual problem is shown in Fig. 5; it is the tendency of most room/loudspeaker combinations to exhibit a substantial peak in the 80 to 200 Hz range with a fall-off of bass response below 60 Hz.

The difficulty is largely one of the architectural acoustics; the majority of living rooms are too small to propagate extremely low frequencies as efficiently as the shorter wave lengths of the mid-bass and lower mid-range regions. The high frequencies are furthermore attenuated due to furniture, carpet, and drapes. The usual sound is described as "thick

Fig. 6—The Altec "Concept EQ" loudspeaker system.

and muddy," without the "impact of really massive bass." Any loudspeaker designed for flat response in an anechoic chamber will suffer the same fate in such an environment.

One interesting practical solution to this problem has been met by the Altec Concept EQ, shown in Fig. 6. This system is designed basically as a *flat system* in terms of its energy output into a uniformly absorptive environment. It is equipped with a 3-control variable equalizer, whose response curves are shown in Fig. 7. Here we show the flexibility of high frequency tailoring and suppression of an 80 to 150 peak in the solid curves; the dotted curves show the range of extreme low-frequency boost. Actually, the equalizer portion of the Concept EQ system is a welcome adjunct to most wide-range systems, and is separately available at a cost of $125.

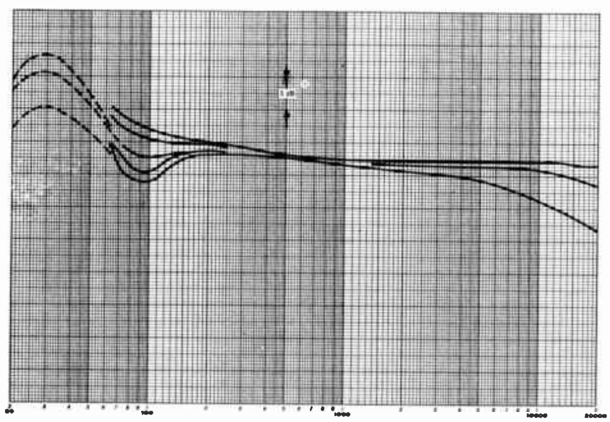


Fig. 7—Electrical response curves typical of the active equalizer portion of the Concept EQ System.

Another very well-known system with an integral active equalizer is the Bose 901. Here, the design aim was to provide the electrical equalization necessary for an array of nine loudspeakers to produce a flat spectral output under the specified mounting conditions of eight speakers facing a reflective wall of large dimensions at a distance of about one foot. The Bose equalizer provides a variety of high-frequency tailoring and a "below-40 Hz" cut-off function for alleviating excessive record rumble and other sources of amplifier overload. There is no specific tailoring of the mid-bass response or adjustment to varying room acoustics. Figure 8 shows the range of the Bose equalizer.

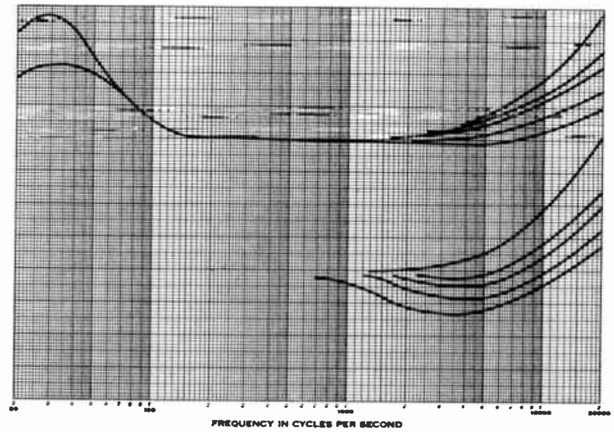


Fig. 8—Electrical response curves of the active equalizer portion of the Bose 901 loudspeaker system.

Hints for the Audiophile

So far, we haven't been too encouraging. We've told you how *good* sound system equalization is—but we've also told you how expensive it is and how difficult it may be to make it perform properly. The well-to-do audiophile in a large urban area can always purchase an Acousta-Voicette for the sum of $875 from a qualified Altec sound contractor or hi-fi dealer and be assured that the equalizer will be properly adjusted.

On a less ambitious level, he can buy an equalizer, either one-third or one-octave, and equalize it himself by ear—or through the use of any of a number of phonograph discs made for the purpose. Such records date back to early issues of the famous CBS Laboratories Test Record Series, and in recent times the Altec Acousta-Voicing test record has certainly been the best known. This record presents carefully calibrated one-third noise bands covering the range of frequencies which can be adjusted by the Acousta-Voicette. The record is designed to be played back *via* the normal RIAA playback response, and monitored with a Sound Level Meter (SLM). With this device, the sound pressure output from each band is carefully noted and charted. The accuracy of the Acousta-Voicing test disc is ±¼ dB from 50 Hz to 2000 Hz and ±1½ dB above 2000 Hz.

An audiophile wishing to equalize his own system with this disc should observe the following:

1. Make sure that the SLM is operating in its *flat* mode—the so-called "C" scale.
2. Make sure that the phonograph cartridge is a flat one—and in excellent condition. The better crop of today's cartridges working into good preamps just about guarantee this. If the cartridge is not *flat*, then the resultant equalization will be "biased," so to speak, in favor of the phonograph cartridge, and other input signals, a tape recorder for example, will be reproduced with an improper contour.

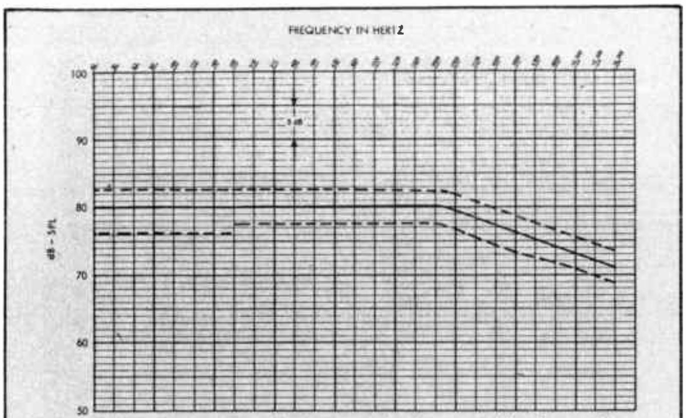


Fig. 9—Optimum acoustical contour for equalization of home systems.

3. Take your time; don't try to equalize too quickly. Adjust a few controls at a time—and then only slightly. Special graph paper is available to facilitate plotting of the curves, and the Altec dealer who can provide the Acousta-Voicette can also provide the disc and the graph paper.

4. Don't try to equalize the system to a *flat* contour. Let it roll off above 2 kHz at about 3 dB-per-octave. This is shown in Fig. 9. A tolerance of $\pm 2\frac{1}{2}$ dB can usually be maintained over the range from about 200 Hz up to 8 or 9 kHz, if sufficient care is taken. Below 200 Hz, the effects of room resonant modes, with their characteristic peaks and dips, tend to dominate the response. In the region below 200 Hz we simply have to accept dips in the response of perhaps 4 dB; peaks should be held as low as possible. Always perform the equalization with the SLM located at the prime listening position.

Why the Response Should Not Be Flat

At least at the present time, the acoustical response of a playback system, whether in the studio or in the home, should exhibit a rolled off response above 1 or 2 kHz. Let us see why this should be the case by going all the way back to the recording studio and examining the consequences of flat monitoring of recorded product. At this point, the author quotes from a paper given at the AES Convention in Rotterdam, February 1973:

"Another important consideration in equalizing monitor systems is the precise tailoring of the high-frequency response. The question of 'flat *vs.* rolled-off' response has been discussed in some detail, and there is a general consensus among recording engineers that some sort of high frequency roll-off is desirable. The reasons, of course, are obvious; most home playback equipment exhibits substantial high-frequency roll-off at normal listener's positions, and a recording monitored and equalized over a system exhibiting the same kind of roll-off will convey most of the musical values the recording producer had in mind. On the other hand, a recording monitored and equalized on a flat system would surely sound dull and lifeless played over the rolled-off system. The answer to the problem is not to make all systems flat; that would call for a reassessment of present disc equalization standards, not to mention the problems of playback equipment design and obsolescence. Rather, the answer is to be found in standardizing on a degree of roll-off, with reasonable tolerances, which can be met by the manufacturers of home playback machinery and studio monitors alike."[1]

1. J. Eargle: "A Summary of Recording Studio Monitoring Problems" given at the 44th AES Convention in Rotterdam, February 1973.

Pre-amplifiers

Notes: (1) All models solid-state except when model number is preceded by (T).
(2) "K" indicates kit price; "W" wired

MANUFACTURER	MODEL	Frequency response, Hz ± dB	Rated output, V	THD at rated output, %	IM at rated output, %	Rated output S/N, phono, dB	Phono sensitivity, mV	Phono overload, mV	Tape-head sens., mV	High-level sens., V	Tape-monitor Z, ohms	Dimensions, W x D x H	Weight, lbs.	Price	SPECIAL FEATURES
ACE AUDIO CO.	Basic Stereo Preamp	20-20k ± 0.1 dB	2.0	.05	.05	-70	2.2	250		0.1	50k	11 x 8 x 3	5	69.95K 99.95W	No tone contl.; flat resp. only; can be used with equalizer.
	Zero-dist. Preamp	DC-267k to −3	1.0	0	0	-65	10.0	110		1.0	50k	11 x 8 x 3	5	69.95K 87.50W	Indent. to above but w/o high-level amps.
ACOUSTICAL MFG.	Quad 33	30-20k ± 0.5	0.5	02		85	2.0	120	*	*		10¼ x 6½ x 3⅝	6½	196.00	*Phono & tape inputs adj. w/plug-in boards
AUDIO RESEARCH	(T) SP-3	5-30k ± 1	5	0.005	0.005	70	2.0	400		0.1	10k	15⅝ x 12 x 5		595.00	Opt. finish on front panel.
BGW SYSTEMS	4XPA 4-chan.	20-20k ± 0.1	10	0.02	0.006	100+	1.0	125		0.1	600	19 x 10 x 7	20	750.	Scope display; CD-4 decoder; mic line inputs avail.; 12 eq. filters.
	2XPA	20-20k ± 0.1	10	0.02	0.006	100+	1.0	125		0.1	600	19 x 10 x 7	20	500.	Can be converted to 4-chan.
CROWN	IC-150	3-100k ± 0.6	2.5	0.05	0.01	90	0.8-8*	33-330*		0.22	600	17 x 8⅛ x 5¼	10	299.00	*Adjustable; wal. cab., $33.00; rack mtg. brackets, $10.; stereo image contl.; 2 tape mon.
DAYTON WRIGHT	SPM	7-50k ± 2	8	0.005	0.01	75	1.2	270		0.4	10k	5½ x 6½ x 10	6	289.00	
	SPG	7-50k ± 2	8	0.004	0.01	80	1.0	250		0.4	10k	17 x 6½ x 10	19	895.00	Invert. & non-invert. outputs; 8 freq. contls; mon. & dial for 2 tape decks, 2 switchable UV mtrs.
	SPX	10-40k ± 2	8	0.007	0.015	75	1.0	270		0.4	10k	13 x 6½ x 10	14	500.00	Built-in comp./exp. sys.
DYNACO	PAT-4	10-100k ± ½	2.0	0.05	0.05	70	3	80	3	0.15	600	13½ x 9 x 4	10	94.95K 159.95W	Front panel input & output; 3-pos. hi filt.
	PAS-3(T)	10-40k ± ½	2	0.05	0.05	70	2	250	2	0.2	47k	13½ x 9 x 4	11	79.95K	3-pos. blend swit.
ESS		2-2000k ± 1	2.5	0.0075	0.005	80	2.5	70		0.3	*	16⅝ x 9 x 6		395.00	Suppressed on/off transients; sq. wave rise time, 50 nano Sec.
EPICURE		10-100k ± 1	5	0.01	0.02	80	1-3	140 250		0.2	5K	19 x 13 x 8			Multi-purpose 'scope; 7-pos. equal.
HARMAN/KARDON	Citation 11	2-250k ± 1	6.0	0.05	0.05	72	2.5	200		*	20k	16⅛ x 12 x 4¾	20	350.00	5 pt. eqlzr.; spkr. swit. w. 'phones; *0.15V, Aux. 1 & 2; 3V., Aux. 3.
INTEGRAL SYSTEMS	10	20-100k ± 0.25	3.5			90	1:4	140; 35		0.2				300.00	
JVC	VP-10	10-100k ± 0.5	3.0	0.03	0.05	84	1.0	120	1.2	0.17	10k	19 x 13½ x 6	22	599.95	7 pos. SEA tone contl.
	VP-100	18-50k ± 0.5	1.0	0.03	0.1	80	1.2	120		0.12	12k	17 x 11½ x 5½	19	259.95	7 pos. SEA tone contl.; pink noise tester.
	4DD-5 4-chan.	20-16k	0.3				1.5					7 x 13 x 3½	5	119.95	CD-4
OLSON	AM 368	30-20k	250 mv.	0.75	0.75	62	2.0	45				4 x 1½ x 2½	1	10.98	
MARANTZ	3300	20-20k ± 0.25	3.0	0.02	0.02	100*	1.25	100		0.135	900	15⅜ x 8¾ x 5¾	22	395.00	*Phono dyn. rng.: ratio of overload to equiv. input noise.
	4000 4-chan.	20-20k ± 0.5	2.1*	0.1	0.1	0.1				0.18	900	15⅜ x 8¾ x 5¾	22	249.95	*Rear; front, 180mV.; 4-chan. preamp adapt.; rear chan. tone contls.; vari-matrix; rem. contl. opt.
PHASE LINEAR	4000	20-20k ± 0.25	2.0	0.25	0.25	82	2.5	120		0.25	600	19 x 10 x 7	18	599.00	
QRK (CCA)	Ultimate I (mono)	50-15k ± 1	1.0	0.2	0.2	-75	12					9 x 3 x 3	2.5	149.50	600 ohm bal. output; built-in rumble filter.
	Ultimate II (stereo)	50-15k ± 1	1.0	0.2	0.2	−75	12					9 x 3 x 3	4¼	194.50	As above.
QUINTESSENCE	Preamp 1	10-100k ± 0.5	7.0	0.025	0.025	70	15	165		0.1	600	14 x 10 x 4	14	350.00	
	Equalizer 1	10-100k ± 0.5	7.0	0.025	0.025	95				0.1	10k	14 x 10 x 4	14	350.00	Act. equal.; ± 12 dB at 120 Hz, 12 kHz shelvg.; ± 12 dB at 400 Hz, 1.2 kHz; 4 kHz peak & dip.
RADFORD (AUDIONICS)	SC24 Mk II	20-50k ± 1	1.0	0.01	0.01	75	2.0	200		0.08	150	16 x 4½ x 8¾	18	429.95	Mid, phone, sensitivity contls.
	HD22	20-50k ± 1	10	0.01	0.01	80	1.0	750		0.08	150	16 x 4½ x 8¾	14	395.00	Two tape inputs.
SAE	Mk IB	10-100k ± 0.25	14	0.02	0.02	80	2.5	100		0.25	5k	17 x 5¾ x 10½	20	750.00	Step contls.; 7-band tape or line out act. eqlzr.; 3-pos tape copy.
	Mk IM	10-100k ± 0.25	14	0.02	0.02	80	2.5	100		0.25	5k	17 x 5¾ x 10½	20	600.00	Dir. rdg. VU mtrs.; no tone contls. Can be used w. eqlzr.
	Mk IX	10-100k ± 0.25	14	0.02	0.02	75	2.5	100		0.25	5k	17 x 7 x 5¾	15	450.00	5-band act. eqlzr.; slide contls.; tape copy.
	Mk XXX	10-100k ± 0.25	9	0.03	0.03	72	2.5	100		0.28	5k	15 x 8 x 4¾	10	250.00	Slide contls.; mil. spec. epoxy crt. brds.
SONY	TA-2000F	10-100k ± 1	1.0	0.03	0.05	73	1.2	300		0.11	10k	15¾ x 12⅞ x 5⅞	19¾	579.50	VU mtrs. made to NAB specs., low noise FET amps.
SOUNDCRAFTSMEN	PRP 18-12	10-100k ± 0.29	3.5	0.02	0.02	85	2.0	110		0.2	600	18 x 12 x 5¼	28	499.50	Comb. Equal. w/9-oct. contl./chan., plus compl. preamp contl. ctr.
	20-12	20-20k ± 0.5	7.0	0.05	0.05	90				0.2	3k	18 x 11 x 5¼	22	299.50	Stereo equal. preamp; freq. bal.
	RP10-12	20-20k ± 0.25	7.0	0.01	0.01	95				0.2	600	18 x 11 x 5¼	22	349.50	Prof. model; switched rec. & PB; input-output level metering.
SOUTHWEST	198	1-80k ± 1	5.0	0.05	0.02	65	2	100		100mV	10k	8 x 12 x 6		69.50K	Pushbutton tone comtrols.; push-pull compl. amp mods.

WHY YOU SHOULD BUY FOUR-CHANNEL NOW

Harry Maynard

Don't buy a four-channel system now, or adapt your current stereo to play four channel because it's new, or just to impress your friends. Buy a four-channel system now because per dollar invested it will give you far better sound than the equivalent invested in adding niceties to your two channel system. Stated simply, $500 or $1,000 invested in a four-channel system playing even your two-channel stereo records will give you better sound than the equivalent investment in stereo sound.

There are certain things a stereo system can't do. It can't properly distribute bass around a 360 degree perimeter, and it can't significantly enhance your existing investment in stereo recordings. Studies show that if you are typical, you have considerably more invested in software (recordings) than you have in hardware. There is also FM radio—FM means free music. FM is also the listening booth for most people who purchase stereo recordings. Most FM stations have record libraries running into the hundreds of thousands of dollars, at your command to decode in the four-channel mode.

U.S. consumers like you have already invested over nine and a half billion dollars in stereo recordings since 1954. With any one of the better matrix four-channel decoders with, of course, two extra speakers and another stereo amplifier, this huge recorded repertoire is at your service for enhancement. Sure, you'll also have the ability to play the growing number of four-channel recordings. But if history is any measure, the transition from stereo to quadraphonic sound will take years. The bulk of all recorded repertoire for many years will be in stereo and often what you will want in recorded sound will only be available in stereo. Some great historical performances will never be recorded in quadraphonic sound. Some great stereo performances that sound dead or inadequate in stereo can, with a good 4-chan. system, be brought back to life to a considerable degree. There is a lot of enjoyment to be mined from your existing recording collection when played in the four channel mode.

Here are some non-expert, authentic reactions on this point addressed to my radio program, "Men of Hi Fi." "After much hesitation and deliberation, I finally purchased an inexpensive third speaker and hooked up the Dynaco system. The result—a new world of stereophonic enjoyment has been opened up for me. The results are nothing less than astounding. Listening to old records is a new experience as if they were being played for the first time. The third speaker has increased my enjoyment by 100 percent. Fantastic!" Or, "I spent many delightful hours rediscovering my record collection."

Hundreds of letters addressed to my show have similar words and reactions from those using the simplest decoders to the most sophisticated matrixed decoders such as the Sony 2020, or the decoder included in the Lafayette LR 4000. Ninety-nine percent of the letters indicate that they now keep their rear speakers on most of the time, even listening to two channel material. Listen—"I have not been able to live with the rear speakers off since I installed the decoder . . . the presence is phenomenal. The Mormon Tabernacle has been airlifted to my basement."

Your reaction now may be, *but these are novitiates in audio.* So what does Julian Hirsch, a dean of audio equipment evaluation of Hirsch-Houck Labs, say? "On almost any kind of stereo material, the EV-4 added a sense of spaciousness that we found most pleasing. In a sense, it was not unlike the Dynaco which adds this quality to many programs. It was interesting to find that there frequently was a definite front/rear separation in ordinary stereo programs, often with a hint of separation between the rear speakers as well. In fact, some normal stereo records sounded at least as good as some of the encoded records!"

This four-channel decoding of stereo records is more than a serendipitous bonus. As Hirsch goes on to say, "We are enjoying playing some of our old forgotten stereo discs and discovering a new dimension of sound hidden in the grooves. Four-channel playback generates a sense of involvement with sound which is so easy to accept that after awhile one may be unaware of its existence. On many occasions we switched off the rear speakers and the contrast was striking. It can only be compared to turning off most of the lights in a well-lit room, after one has become adjusted to a high ambient light level."

Finally, like many leading audio critics, Julian says what so many listeners have written to me on my radio show, "We had been highly skeptical of early claims that four-channel sound was as much of an improvement over two channels as the latter was over mono. At this point we are ready to eat crow. Going back to mono is an intolerable prospect for a real stereo addict."

For knowledgeable people in audio it's been an open secret that many stereo records have been matrixed for years. Even the so-called discrete CD-4 records are matrixed in the recording process and, incidentally, decode nicely with a matrix decoder. When you encode a four-channel recording, you are primarily making more calculated, and sometimes more satisfactory and efficient use of the two-sided groove wall of a stereo disc by more organized use of rear phase material.

I have often found that the latest generation of matrixed decoders, with their front-to-back logic and full wave matching logic, on a stereo record significantly improve front-to-back separation and give more precise localization of voice and instrument, as well as increase separation, which was precisely what they were intended to do with an especially encoded four-channel record.

Now the purist will complain at this point that what we are hearing out of the rear speakers was never intended by the artist, record producer, and recording engineer. True. Perhaps if I were mixing this stereo record for four-channel listening, and this includes some especially encoded four-channel recordings, I might not have placed the instruments as they are placed or aimed for the total effect of this recording. But generally, the effect is definitely more pleasureable, and of course I can always turn off the rear speakers if I find it really offensive to my sensibilities.

But I have found as hundreds of my listeners have that I keep my rear speakers on 99 percent of the time. Let some of my listeners describe their reactions: "It adds a great deal of excitement to music, which for myself, stereo did not," or, "During the listening (I) shut off the two rear speakers for a little while. No one in his right mind could fail to notice the difference. Then (I) put all four speakers on again. The sound seems flat and dead by comparison. With four channel sound we are living in a new world that Toscanini, Caruso, yes and even Paganini, would have loved. Let's appreciate the beautiful sound we now can enjoy (from stereo discs) until the F.C.C. decides that discrete is better."

For me, the last sentence of this letter indicates why the discrete-matrixed debate is for all practical purposes not as meaningful a debate as it appears, and has tended to muddy the water of today's enjoyment. I believe that both systems will exist side by side for years, but that is the subject of another piece. Suffice it to say that the matrixed four-channel disc is the natural evolution of the stereo disc. Any good four-channel system needs four-channel matrix decoding facilities to enhance two-channel stereo records, because stereo records for now and for the near future constitute the bulk of recorded repertoire available to the public. It took fifteen years for the mono/stereo shift. It will take years for the stereo/quad shift. Yes, I know four-channel recordings have got off to a much faster start than stereo did, in a far shorter time period.

Right now, I'll even grant the discrete camp that I find discrete tapes and discrete discs sound better than the best of the matrix system commercially available. But I've heard the laboratory prototype decoders of the two main matrix camps, Q.S. (Sansui) and S.Q. (Columbia), and I've been impressed (with no encoding changes) with their tremendous improvements in separation and other criteria. I can't tell the difference between the master four-channel tapes and the matrix decoded material. These prototype models will be converted to IC chips by late 1973 and 1974.

Most informed observers of the four-channel scene agree that we won't have an F.C.C. approved form of four-channel discrete broadcasting for several years. It's taken fifteen years of stereo broadcasting just to get one-third of America's FM stations to go stereo, partly because of the cost involved. Four channel will require an equivalent or bigger investment by radio stations. Since by now the reader has obviously gotten the point that I believe the enhancement effect alone on stereo records is sufficient reason for setting up for four-channel listening. From where I sit, the proponents of discrete four-channel sound should have invented matrixed four channel sound until discrete four channel can be broadcast. Matrix four-channel sound is the natural bridge from the stereo age into the quadraphonic age.

One thing I'm sure of—four-channel sound is not a put-on. If anything, it's been too long in arriving in the commercial market place (subject for another article). It's certainly here to stay with retailers such as Lafayette, the nation's second largest hi fi component dealer, reporting that 60 percent of their total component hi fi sales is in four-channel equipment, and other dealers reporting a real upsurge in sales in four-channel recordings and equipment.

But if four channel has a natural resistance point, much research by this reporter shows that resistance point and fear is OBSOLESCENCE of both the consumer investments in software (recordings) and in hardware. People hate to hear that the recordings and equipment they have carefully acquired, grown to love, and invested much money in, now have to be thrown out. It you don't believe this, stand and overhear the hundreds of conversations I've heard, or read the mail addressed to my radio program. For example, here's a letter addressed to Jim Gabbert, one of four channel's pioneer broadcasters, the head of the National Association of FM Broadcasters engineering committee and editor of its newsletter, *FM Engineering*, "Well I see it (four-channel) is on its way in all its glory. Equipment manufacturers are drooling like mad dogs on the Fourth of July, while their greedy little heads whirl at the thought of those dollars rolling in from stupid suckers who will buy anything if it costs money. I refer to your report on quadrasonic, quadraphonic quadrout, quadrafool, quadraput-on, surround sound as described in your newsletter."

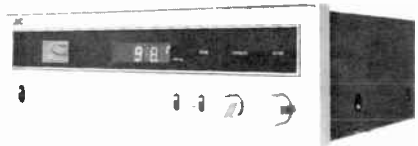
"I can just see it now—broadcast antennas, new transmitters, new control consoles, new cartridges, new home hi fi systems, complete with infinite baffle speakers built into all walls of every house (which calls for new houses). It staggers the imagination when one considers how easily Yankee ingenuity has made all present audio equipment (from which incidentally manufacturers made their fat little profit) OBSOLETE!"

For AUDIO's readers I don't think it's necessary to refute this letter in detail. Many readers could, I'm sure, do a better job than I, knowing that most of your investment in quad if you already have a good stereo system, is add-on equipment, i.e. another stereo rear amp, and two rear speakers, plus whatever level of matrix decoder you desire. Nothing is made obsolete. Most decoders or four-channel systems (of audio component quality) now being offered to the public have auxiliary four-channel inputs which can be used to play four-channel tapes, four-channel tape cartridges, and even add a demodulator (price $100) to play the new CD-4 records.

To sum up, don't forget what a four-channel system will do for those unforgettable stereo records you own, or may still buy. Martin Mayer, *Esquire's* audio-record critic, a recent convert to four-channel sound, suggests as I do that with all the shouting about what's new about four-channel, "the advantages of four-channel sources even on ordinary stereo material is much greater than one could imagine without trying it. The experience of Janos Starker playing "Bach Cello Suites" (a two-channel recording) through four cornered omnidirectional speakers was a great musical moment in my house, because it was indistinguishable from what one would hear if the artist were playing in the room. The sound of the live cello in a room does not seem to come from a point source or from the front wall. The whole room plays . . . the whole room resonates. No area is louder than the other so far as you can tell." I remind you that Mayer's reaction is to a two-channel recording played quadraphonically.

Mayer's experience is not atypical for those who have lived with four-channel stereo. It's confirmed by thousands of my radio listeners and lecture audiences who have told me essentially the same thing. So if you want to double the sonic value of many of the recordings you already own, and step up your psychic income, invest in four-channel now. There is a lot of four-channel gold in them thar two-channel stereo recordings that cannot be mined with your existing two-channel stereo equipment, no matter how much you spend. For the added investment needed to convert your two-channel system to four-channel, you'll get a lot more sonic value than the equivalent investment in two-channel niceties.

Heath AJ-1510



JVC VT-900

Kenwood KT-8005

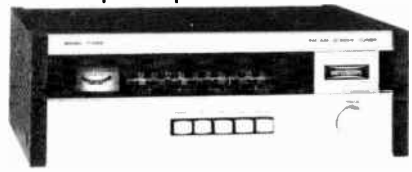
Notes: (1) All models solid-state except when model number is preceded by (T).
(2) "K" indicates kit price; "W" wired.

MANUFACTURER	MODEL	IHF sensitivity, µV	Capture ratio, dB	AM suppression, dB	Alt. chan. selectivity, dB	Frequency response, Hz ? dB	Stereo separation, 1000 Hz dB	Stereo separation, 10 kHz dB	THD, mono, 100% mod., %	THD, stereo, 100% mod., %	Tuning indicator	S/N, dB	Number of meters	AM band?	Dimensions, w x d x h, in.	Weight, lbs.	Price	SPECIAL FEATURES
ACOUSTICAL MFG.	Quad FM3	1.0	3	46		20-15k ±1	40	25	0.3	0.5		65		no	10¼ x 6½ x 3⅝	6	237.00	Quad twin lamp tuning sys.
ALTEC	724A	1.8	1.3			20-15k ±1	40		0.3		mtr		2	yes	17¾ x 16½ x 5	34	499.00	Incl. preamp; phono sens. 2.0mV or 5.0mV switchbl; S/N: 80dB, hi lev.; 60dB, lo lev.
DYNACO	AF-6	1.75	1.5	65	58	50-15k ±1	40	30	0.5	0.5	lt., mtr	65	1	yes	13½ x 11 x 4	13	225.00K 325.00W	Dynatune auto. tuning cirt.
	FM 5	1.75	1.5	65	58	50-15k ±1	40	30	0.5	0.5	lt., mtr	65	1	no	13½ x 9 x 4	11	159.95K 249.95W	Same as above.
HARMAN KARDON	Citation 14	2.0	1.5	60	60	4-80k ±1	45	35	0.25	0.35	2 mtrs*	70	2	no	16 x 13¾ x 4⅞	27	525.00	*"Quieting" mtr. (pat.); PLL demod.; Dolby decod.; test outputs; 4-chan. output.
	Citation 15	2.0	1.5	60	60	4-80k ±1	45	35	0.25	0.35	2 mtrs*	70	2	no	16 x 13¾ x 4⅞	27	395.00	"Quieting" mtr. (pat.); PLL demod.; 4-chan. output.
HEATH	AJ-14	5.0	3		35	20-15k −3	30		1.0			50		no	12 x 9⅜ x 3	4¼	49.95K	Preassembled FM front-end.
	AJ-1214	2.0	2	60	50	20-15k ±1	40		0.5	1.0		65		yes	12¾ x 13 x 3⅞	7⅜	89.95K	Same as above w/ FET RF; ceramic filters; PLL mpx.
	AJ-29	1.8	1.5	70	50	20-15k ±1	40	30	0.5	0.5	mtr	60	2	yes	16¾ x 4½ x 5⅛	14½	179.95K	Mod. constr.; preassem. FM front end w/FET RF; 9-pole L-C filt.; 3-FM-i.f. ICs; AM w/adj. rod antenna.
	AJ-15	1.8	1.5	70	50	20-15k ±1	40	25	0.5	1.0	mtr	65	2	no	16⅞ x 12½ x 4⅜	11½	199.95K	Preassem. FET FM front-end; x-tal filts.; noise oper. squelch; stereo only swit.; 2 stereo 'phone jacks w/lev. contls.
	AJ-1510	1.8	1.5	95	60	20-15k ±1	40	30	0.3	0.35	*	65	1	no	16⅜ x 14¾ x 6	16	539.95K	*PLL tune & MPX; 4 digit readout; sweep, card, & push button tune.
HITACHI	FT-600	1.8	1.5	45	50		40			0.8	mtr	70	2	yes	16 x 5 x 12	19	229.95	
JVC	VT-900	1.7	0.8	70	55		38	25	0.3	0.5	mtr*	65	mtr*	no	16⅝ x 12½ x 5½	19½	369.95	*Dig. readout; IC & FET; dual element FM filters.
	VT-700	1.7	0.8	70	55		35	25	0.3	0.5	2 mtrs	65	2 mtrs	yes	17 x 12 x 5½	16½	299.95	4 FM mech. filt.; mpx filter.
KENWOOD	KT-8005	1.5	1.0	100	65	50-15k ±0.5	40	40	0.2	0.3	mtr	75	2	yes	17⅛ x 11¾ x 6	28.4	389.95	4 FET, 5-gang front-end; ICs & four 2-element filt. in i.f.; dbl.-swit. demod.
	KT-6005	1.5	1.3	80	60	20-15k +0.2, −1.0	45	38	0.3	0.5	mtr	70	2	yes	17⅛ x 11¾ x 6	17.8	289.95	Dbl.-swit demod. (DSD); FET front-end; mpx filter; linear dial.
	KT-4005	1.9	2.0	60	55	20-15k +0.2, −1.5	40	35	0.4	0.7	mtr	70	2	yes	17⅛ x 11¾ x 6	17.8	189.95	Dbl.-swit. demod.; FET front-end; IC i.f. stg.; mpx filter; muting.
	KT-2001A	2.0	4.0	45	45		30	20	0.5	0.7	mtr	60	1	yes	13¾ x 9⅛ x 4⅝	10	119.95	FET, 3-gang front-end; mpx filter.
LAFAYETTE	LT-725A	1.7	1.5	50		50-15k	40		0.25		mtr	75	2	yes	12 x 9⅛ x 3¾	12¼	149.95	Mute; tape out; internal FM antenna.
	LT-670B 4-chan.	3.5	5	35		50-15k	30		0.7		lt.	50		yes	10⅝ x 8⅜ x 3½	12	107.50	4-chan. output jack; AFC; MPX filter swit.; built-in antennas.
	LT-D10	1.65	1.5	60	45	50-15k ±1	40		0.2		mtr	70	2	yes	15⅜ x 11⅞ x 4½	11	229.95	Dolby noise reduc. & de-emphas.; front, rear tape outs; PLL MPX; MPX hi filt & mute.
LEAK (ERCONA)	Stereofetic FM tuner	2.0	3.5	55	50		30	20		0.25	mtr	60	1	no	11½ x 4¼ x 7¾	6	225.00	
MARANTZ	120	1.4	1.5	80	60	30-15k ±1	42	30	0.15	0.25	'scope	70		yes	15⅜ x 13 x 5¾	29.4	429.00	'Scope display for tuning, multipath audio & ext. src.; FM quadradial output (detector output).
	115B	1.9	1.5	60	50	30-15k ±1	42	28	0.3	0.4	mtr	70	2	yes	15⅜ x 13 x 5¾	24	279.95	PLL mpx demod.; FM quadradial output.
	105	3.5	3.0	48	55	30-15k ±2	40	25	0.6	0.8	mtr	55	1	yes	14¼ x 12 x 4¾	20	149.95	
OLSON	RA-310	3.5	0.9	3.8	35	50-15k +3 −3	37	28	1.6	1.8	mtr	58	1	yes	11½ x 7¼ x 4¼	10	69.99	

Tuners

Pioneer TX-9100

Sansui TU-7500



Superscope T-220

Notes: (1) All models solid-state except when model number is preceded by (T).
(2) "K" indicates kit price; "W" wired.

MANUFACTURER	MODEL	IHF sensitivity, µV	Capture ratio, dB	Alt. chan. selectivity, dB	AM suppression, dB	Frequency response, Hz ± ? dB	Stereo separation, 1000 Hz dB	Stereo separation, 10 kHz dB	THD, mono, 100% mod., %	THD, stereo, 100% mod., %	Tuning indicator	S/N, dB	Number of meters	AM band?	Dimensions, w x d x h, in.	Weight, lbs.	Price	SPECIAL FEATURES
ONKYO	T-4055	1.7	1.2	80	55	20-15k +0,−2	40	35	0.2	0.5	mtr	70	2	yes	16⅝ x 14 x 5⅜	20	199.95	Audible multipath dist. contrl.; 1.7µv sens.
PILOT	211	1.8	1.5		65	20-15k	36		0.4	0.8	mtr	65	2	yes	15 x 11½ x 5⅛		199.90	Mute; MPX filter 4 gang FM tuning.
PIONEER	TX-6200	1.9	1.5	60	50	20-15k +0.2,−2	40	30	0.2	0.4	mtr	70	1	yes	17 x 13 x 5¼	15	139.95	Muting; mpx filter; fixed & var. output levels; FM multipath output.
	TX-7100	1.9	1.0	60	55	20-15k +0.2,−2	40	30	0.2	0.4	mtr	70	2	yes	17 x 13 x 5½	17	199.95	Output lev. AM & FM: mpx noise fil.; muting; multipath output; fixed & var. output level contls.
	TX-8100	1.8	1.0	80	55	20-15k ±2	40	30	0.2	0.4	mtr	70	2	yes	17 x 13 x 5½	17	249.95	Output lev. contls. for phones, AM & FM modes; mpx noise fil.; muting; multipath output; fixed & var. output.
	TX-9100	1.5	1.0	90	65	20-15k +0.2,−2	40	30	0.2	0.3	mtr	75	2	yes	17 x 13 x 5½	19	349.95	Output lev. contls for phones, AM & FM modes; pulse noise sup. swit.; 2 lev. mut. swit.; mpx noise fil.; multipath outputs, output fixed & var.
RADFORD AUDIONICS	FMT-4/5	1.2	1	100	60	30-15k ±1	45	40	0.2	0.5	mtr	70	2	no	16 x 4½ x 8¾	20	475.00	Avail. in 2 exter. finishes to match MK I or Mk II series.
RADIO SHACK	TM-90	4	2.5			20-20k ±2	34	20		0.3	mtr	55	1	yes	14½ x 11 x 3½	13	119.95	FET front-end; wood case.
	TM-175B	5	2.5			20-20k ±2		25		0.3	mtr	48	1	yes	7 x 10 x 4	8	79.95	FET front-end; wood case.
	TM-100	5	3.0			20-20k ±2	25	15		0.3		40	0	yes	9 x 3 x 6½	7	49.95	Wood case.
REVOX	A76	1.0	1.0	80	54	30-15k ±1	40	30-		0.2	mtr	70	2	no	16⅜ x 9⅝ x 6¼	18	599.00	Var. trigger lev.; multipath ind.; ctr. tuning mtr.; sig. str. mtr.
ROTEL	RT-222	4.0	3	20			32			1.0	mtr	63	1	yes	11¾ x 7⅛ x 4	7	109.95	Wood cabinet.
	RT-322	2.0	2.5	55			35			1.0	mtr	65	1	yes	12⅜ x 8¾ x 4	9	159.95	Mplx. filter; FM muting.
	RT-622	1.7	1.5	60			40			0.9	mtr	65	2	yes	16½ x 12 x 5½	11	239.95	Mplx. filter; hi & lo level outputs; adj. muting level.
	RT-1210	1.5	1.0	90			46			0.5	mtr	70	2	yes	16½ x 12 x 5½	17	299.95	Built-in preamp; 3-pos. FM muting.
SAE	MkVI	1.6	1.5	140	100	20-15k ±0.5	50	38	0.1	0.15	*	75		no	17 x 10½ x 5¾	25	1050.00	*Dig. readout: 3" 'scope; 14 pole filt.; 4 ganged FET front end.
SANSUI	TU-9500	1.7	1.5	80		30-15k +0.5,−2	40	30	0.2	0.3	mtr	75	2	yes	19⅝ x 13⅝ x 5½	21	299.95	
	TU-7500	1.9	2	70		30-15k +0.5,−2.5	40		0.3	0.5	mtr	70	2	yes	17⅜ x 12⅝ x 5½	17¾	249.95	
SEQUERRA	1	2.0	*	130	70	20-15k ±0.2	50	36	0.1	0.2	**	80	0	no	16 x 14¼ x 5 3/16	30	1800.00	*Co. does not pub. fig.;**'scope 4½" flat face; panoramic spect. analyzer, $400.; rem. pushbtn. tuning; 4-chan. 'scope display.
SHERWOOD	S-2400	1.8	1.5	65	60	20-15k ±1	40	25	0.25	0.5	2 mtrs	70	2	yes	17½ x 14 x 5	29	229.95	Wal. case; 'scope outputs; var. output; 4-chan. FM out.
	SEL300	1.5	1.7	80	65	20-15k ±1	40	30	0.15	0.25	2 mtrs	70	2	no	16¼ x 14 x 5¼	25	499.00	Dig. readout; 'scope outputs; tape mon.; tape dubbing; 'phone amp; 4-chan. FM out.
SINCLAIR (HERVIC)	2000	2.0	1.5				40		*	*	2 mtrs.	65	2 lts	no	12 x 6 x 2	3	129.50	*0.15% THD 30% mod.; var. tuning; rem. contl.; PLL disc.; anod. blk. alum. case; 3000 same w/silver anod. alum. case.
SONY	ST-5055	2.2	1.0	70	45	30-15k ±2	35		0.4	0.6	mtr	68	1	yes	16¼ x 11¼ x 4¾	10¾	189.50	FET front-end. linear FM dial scale; wood cab.
	ST-5150	2.0	1.0	70	56	20-15k ±1	40		0.3	0.5	mtr	70	2	yes	15¾ x 13⅝ x 5⅞	15¼	269.50	FET front-end; 75 ohm antenna "F" conn., linear FM dial scale.
	ST-5130	1.5	1.0	100	60	20-15k ±1	42		0.2	0.3	mtr	75	2	yes	15¾ x 13⅝ x 5⅞	16½	369.50	Same as above plus INS impulse noise suppressor.
SUPERSCOPE	T-210	5.0	6	35		30-15k ±3	30	15			mtr	60	1	yes	14½ x 8 x 4½	5.3	99.95	Sig. strength tun. mtr.; bal. flywheel tuning; stereo ind.; AFC for FM.
	T-220	2.8	3	35		20-15k ±1.5	32	20			mtr		1	yes	20½ x 19 x 12¾	18.7	159.95	Oversize tuning mtr.; ill. ind.; mono/stereo swit.; FM muting swit.; 4-chan. FM output jack; adj. output lev.

NEW SPECS FOR THE NEW TUNERS

Leonard Feldman

NEARLY FIFTEEN YEARS ago, the fledgling Institute of High Fidelity undertook what seemed like an impossible task—the formulation of measurement standards for FM and AM tuners. Until that time, most manufacturers published performance specifications based upon standards that had been issued by the Institute of Radio Engineers (now the IEEE) in 1947, just after the birth of commercial FM broadcasting on a national scale. High fidelity component manufacturers realized that the tuner of 1958 could no longer be judged by measurements devised in 1947. To the everlasting credit of a few brave engineers affiliated with what was then a very tiny segment of the home entertainment electronic industry, the "new" standards were issued in late 1958. Before long, everyone who hoped to sell high quality FM receiving equipment adopted the newer standards and a relative measure of specification standardization prevailed.

If you can remember 1958 clearly, you'll recall that in that year, FM stereo was just a dream in the mind of Murray Crosby. Transistors had been invented a decade or so earlier, but were to be found only in a few, expensive imported AM "pocket" radios which presupposed your having pretty large pockets. In short, the "new" tuner specs of 1958 are pretty well outdated and it's time for their retirement—or rejuvenation, if you just can't bear to see "old specs die". As of this writing, the IHF has again turned to the matter of measurement standards and, hopefully, it won't be long before updated standards will be proposed and, accepted by the industry at large. As one who has had to deal with the "old" specs both in my daily work and in the course of reviewing the performance of various FM products for readers of AUDIO, I'd like to propose a few ideas for consideration by the newly formed committee and, perhaps, come up with a set of more meaningful specifications in the light of today's state-of-the-art FM product.

Down With IHF Sensitivity!

If you read FM tuner spec sheets as I do, you may have wondered whether or not there is really a meaningful, *audible* difference between a tuner that boasts, say 1.9 μV IHF sensitivity and one that shamefacedly admits to only 2.0 μV. The fact is the *neither* product sounds particularly good when fed with its "least usable sensitivity" signal (a once-popular synonym for "IHF sensitivity"). That is because with such a minimal signal applied to the antenna terminals, the total noise and distortion content of the resultant audio output will be some 3%. If this percentage consists primarily of distortion (THD), you'd hardly want to listen to the program. If the content is primarily noise—why, even a cheap cassette machine (the garden variety "dictating machine" type) can provide a better signal-to-noise ratio than *that*. Your low-level phono-preamplifier nearly always boasts at least 60 dB of S/N and high-level inputs of amplifiers are capable of 75 or even more dB of dynamic range above noise level. Why worry, then, about whether it takes 1.8 or 1.9 μV of signal input to produce a "listenable" signal that you wouldn't want to listen to?

Usable Sensitivity

Suppose that instead of "least usable sensitivity" we propose that manufacturers specify the least signal which will provide a 50 dB signal-to-noise ratio. Such an S/N is quite listenable, with just a bare trace of background noise. A comparison between this proposed spec and the IHF sensitivity for a given receiver is shown in Fig. 1. The new specification could be stated as: *Usable Sensitivity (50 dB S/N)*: 4.5 μV. Since THD and noise (both unwanted signal content) were correctly linked together by the creators of the original IHF specs, an accompanying specification should be: *THD at Usable Sensitivity*. The number will always be less than 3% and, in our example of Fig. 1, turns out to be a very listenable 1.2%. We're all for continuing the practice of quoting "ultimate" S/N, but feel that instead of arbitrarily measuring it at 1000 μV signal input, some statement should be made establishing just what the signal input has to be before this "ultimate" or best S/N is reached. *That* kind of statement would really separate the "men from the boys". Thus, a typical specification might read: *Ultimate Signal-to-Noise:* 70 dB at 75 μV input and you'd know that the set has a better, faster quieting characteristic than one which states: *Ultimate S/N:* 70 dB at 150 μV. This idea is conveyed in Fig. 1 as well and the same approach could be used to define lowest harmonic distortion (THD), which, in our example would be stated as: 0.3% with 40 μV input.

THD vs. Frequency

Although the old IHF specs rightly recognized the fact that the de-emphasis curve built into the tuner output circuitry tends to make THD readings above 1 kHz or so look "better" than they would otherwise be, we find an increasing tendency on the part of many quality manufacturers to quote THD for frequencies above and below the nominal 1000 Hz and believe that this is a trend in the right direction. Thus, in our new specs, we'd propose that THD be quoted for 100 Hz, 1 kHz and 10 kHz as a minimum, with a full plot (as shown in Fig. 2) optional—but desirable. You're probably not

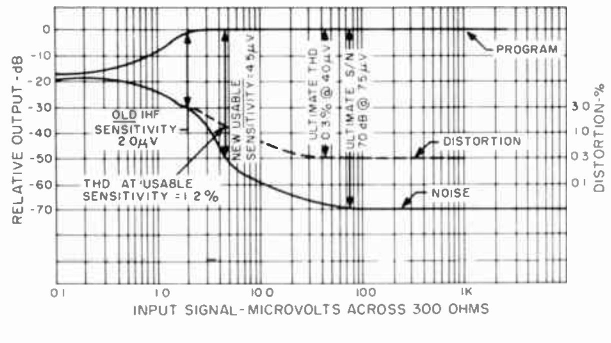


Fig. 1—New, more meaningful FM mono performance specifications include usable sensitivity, THD at usable sensitivity, ultimate THD and ultimate S/N.

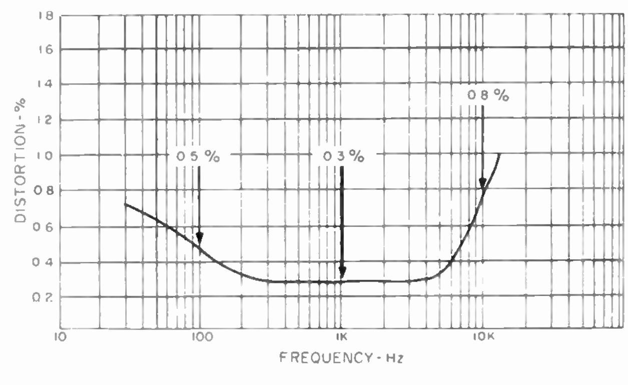


Fig. 2—FM Mono THD versus frequency. Standards would require statement of THD at 3 key frequencies.

too surprised to find that in our example, THD tends to be *higher* at the extremes of the audio spectrum—deemphasis notwithstanding. It wouldn't be a bad idea if manufacturers were asked to specify nominal THD at some high signal level—such as a volt or more, since it is well known that *some* input circuits can be overloaded with disastrous effects as far as distortion is concerned.

All of the other specs relating to mono performance are quite acceptable, as far as we're concerned, and they are tabulated at the end of this discussion for you to examine. Before going on to stereo-related specs (of which there are *none* in the existing pre-stereo standards), one brief word about AM suppression. The true severity of the problem of "multipath signal reflections" in terms of their effect on FM and stereo FM listening has only recently been studied in detail by the FM technical fraternity. Admittedly, AM suppression has been a recognized "spec" all along, but again, it is only measured with reference to a signal level of 1000 μV—and that's not always available for every FM listener. A better approach would be to quote AM suppression referenced to the new "Usable Signal" level as well as to the usual 1000 μV reference. Still better would be a continuous plot of this characteristic, as shown in Fig. 3. Multipath interference often creates AM signal components and the ability of a tuner to reject such components is directly related to how multipath reflections are likely to affect your FM (and especially your FM stereo reception. The new spec would read: *AM Suppression:* 50 dB at Usable Input; 65 dB at 1000 μV input.

Specs For FM Stereo

Most high fidelity component manufacturers, having no guidance from some central authority, have created a set of stereo FM specs to define performance of this portion of their circuitry. Usually included are: "Stereo Separation at 1 kHz" and, more often than not these days, "THD, Stereo at 1 kHz". While we don't think that extremely high orders of separation are that important for a satisfactory stereo localization effect, a more complete specification should, minimally, include separation capability at 100 Hz and 10 kHz. In equipment reviews published in AUDIO, we always show a plot of separation versus frequency, as illustrated in Fig. 4.

Stereo Usable Sensitivity

It is by this time well known that signal-to-noise ratios for stereo FM listening are inferior to those typical in monophonic performance. The reasons have to do with the nature of the composite signal and its wider bandwidth, but anyone who has switched a noisy stereo FM program back to mono knows that mono produces inherently quieter background noise. Some manufacturers "let the noise fall where it may" when the mode switch is set to "Stereo FM". Others only permit the set to "switch to stereo" when a reasonably strong input signal is provided at the antenna terminals. In either case, a specification calling out "Usable Stereo FM Sensitivity" is needed, and its parameters could be exactly those applied to the Usable Sensitivity described earlier—the signal required to produce 50 dB of S/N in the presence of a stereo FM broadcast. A THD specification referenced to this Usable Stereo Sensitivity should accompany this primary spec as well. Thus, the two specifications might read: *Usable Stereo Sensitivity (50 dB S/N):* 10 μV. *THD at Usable Stereo Sensitivity:* 1.7%. For the manufacturer who really wants to go all out, a graph completely analogous to Fig. 1, such as that shown in Fig. 5, could be presented. This graph would also

If *all* of my suggestions were followed and incorporated in a new set of IHF Tuner Standards, a future "spec sheet" for the tuner of 1974 might well look like this:

Monophonic Performance

Usable Sensitivity (50 dB S/N):*	4.5 μV
THD at Usable Sensitivity:*	1.2%
Ultimate S/N:*	70 dB at 75 μV input
Ultimate THD:	
1 kHz:	0.3% at 40 μV input*
100 Hz*	0.5% at 40 μV input*
10 kHz*	0.8% at 40 μV input*
AM Suppression:	
At Usable Input:*	50 dB
At 1 kμV input*	65 dB
Selectivity:	60 dB
Capture Ratio:	1.9 dB
Image Rejection:	85 dB
IF Rejection:	85 dB
Spurious Response Rejection:	90 dB
Frequency Response:	30 Hz to 15 kHz ±0.5 dB
Frequency Stability:	±1 kHz
Input impedance, antenna:	300 ohm bal.; 75 ohm unbal.

Stereo Performance

Usable Stereo Sensitivity (50 dB S/N):*	10 μV
THD at Usable Stereo Sensitivity:*	1.7%
Ultimate Stereo THD:	
1 kHz*	0.9%
100 Hz*	1.3%
5 kHz*	2.4%
Stereo Threshold*	Variable; 7 μV to 20 μV
Stereo Separation:	
1 kHz*	40 dB
100 Hz*	33 dB
10 kHz*	28 dB
SCA Rejection:*	60 dB
Sub-Carrier Product Rejection:*	55 dB

* Items which are not now in the IHF Standards for Tuner Measurement.

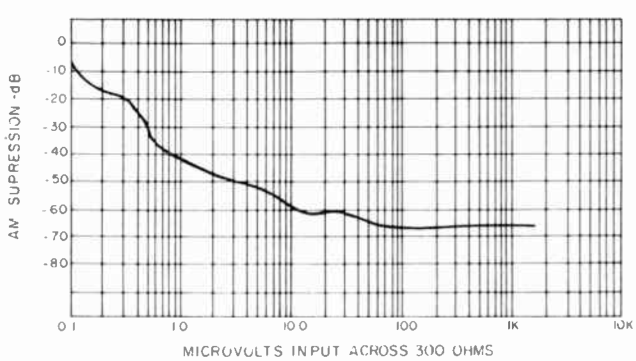


Fig. 3—Relative importance of AM suppression in FM equipment should be stressed in new standards.

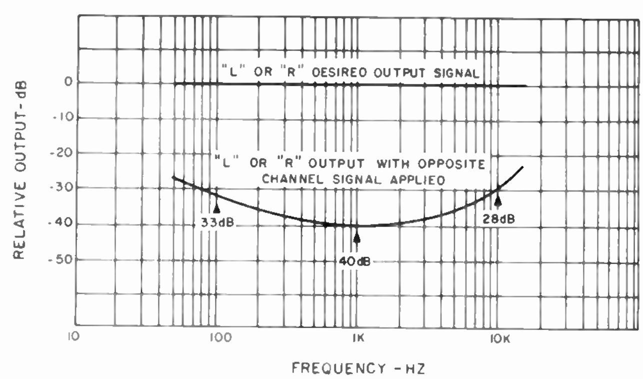


Fig. 4—Separation for FM stereo should be stated in new specifications for at least three frequencies.

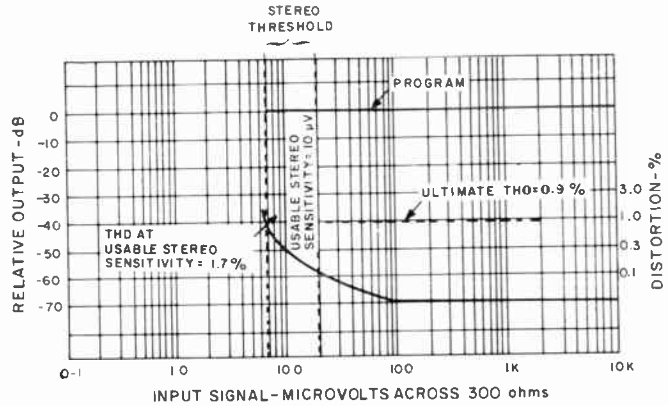


Fig. 5—FM stereo performance characteristics should be plotted or stated separately in new standards.

define "Stereo Threshold" level for such tuners or receivers that switch from mono to stereo at a pre-determined signal level. Since there are some sets which provide a variable adjustment for this switching, the range of adjustment could be easily shown in the same graph. Often, residual 19 kHz, 38 kHz and harmonically related products appear in the outputs of a stereo tuner or receiver. Their level should be quoted separately (and is already, by a goodly number of manufacturers), but if it is less than 50 dB down from program level at 100% modulation, the manufacturer should be permitted to insert a low-pass (15 kHz cut-off) filter when making his Usable Stereo Sensitivity and THD measurements, since these components, while objectionable, are not truly "noise" or "distortion" in the traditional sense and do not contribute directly to our audible displeasure.

Ordinary THD at various frequencies, with the set operating in the stereo mode, poses certain problems which are not present when THD is measured in the mono mode. Because of the presence of super-audible frequencies in the multiplex decoder section of an FM tuner or receiver (19 kHz, 38 kHz, etc.), any non-linearities in the circuits will result in the generation of "beat" frequencies between the desired audio program tone and one or more of these subcarrier high frequencies. For example, a desired 12 kHz tone beating with the locally generated 19 kHz pilot carrier signal may result in a 7 kHz difference signal. A variety of other mathematical combinations are possible and do, in fact, occur in varying degrees when static THD measurements are attempted. The ordinary distortion analyzer cannot discriminate between these extraneous signals and true harmonic distortion and, as a result, some very high "THD readings" may be obtained when specific frequencies are used to modulate the FM test generator in stereo THD measurements. Audibly, these "beats" never seem to sound quite as bad as they appear in static tests—probably because their duration is so short under true musical program conditions. Nevertheless, we do feel that a graphic plot, or at least a statement of THD versus frequency for 100 Hz, 1 kHz and 5 kHz should be included in the new stereo test specifications. 5 kHz is chosen in this instance simply because the measurable "beats" discussed tend to occur when higher frequencies are applied and should not affect true THD readings if they are limited to the frequencies just selected. A full graphic plot is shown in Fig. 6.

A statement regarding the presence of residual subcarrier products at the output is sometimes given by manufacturers and, in our opinion, this specification should become mandatory, since the presence of significant amounts of 19 kHz, 38 kHz or harmonics can seriously affect tape recordings made "off the air". Bias oscillators used in tape recording products also operate at super-audible frequencies and may "beat" with, say, any residual 76 kHz present in the output of the tuner or receiver. While the individual high frequencies may not be audible in the recording, the "beats" themselves may well fall within the audible range and can destroy a well-planned and otherwise flawlessly executed recording.

Tuning Accuracy and Drift

Drift of an FM set after warm-up was included in the old IHF tuner standards and, for many years, most manufacturers included this specification in their brochures. More recently, however, solid state designs (and the attendant reduction in component heating) has reduced overall drift in most of the better tuners to very low values. As a result, most manufacturers have eliminated this specification entirely. In all fairness to those FM products which have lately appeared and which employ phase-lock-loop circuitry referenced to a highly accurate crystal controlled oscillator, we feel that a statement regarding drift should be required once more—if only to point up and justify the extra design cost inherent in the more sophisticated phase-lock-loop approach to tuning accuracy. In the case of these new tuners the drift figure might well be measured in Hz, rather than kHz, but that would provide an opportunity for the prospective purchaser to see just how stable the new products really are.

It is, of course, impossible to predict the course of future FM developments today any more than the writers of the 1958 Standards could have foreseen the need for the changes and additions recommended in this article. Undoubtedly, a means for multi-channel broadcasting (quadraphonic or, perhaps, even more channels) will eventually be sanctioned by the FCC. Considerations of this problem are only just beginning, and it is likely to be some time before we are presented with a new composite signal that may require still newer specifications and measuring techniques. Nevertheless, to wait until those deliberations are concluded would, in my opinion, be a serious error, since technological advancement is a never ending thing and the standards we approve this year, or next, are likely to need revision at some time in the future regardless of how intelligently we write them now.

The modifications for the FM Standards just discussed are by no means all-inclusive. As the new committees are formed, other suggestions are likely to emerge and, for that matter, any readers who have ideas relative to new FM Measurement Standards are invited to send them along to the Institute of High Fidelity, at 516 5th Avenue, New York, N.Y. 10036, attention: Chairman, Standards Committee.

Len Feldman is now Chairman of the above committee—a most appropriate choice-Ed.

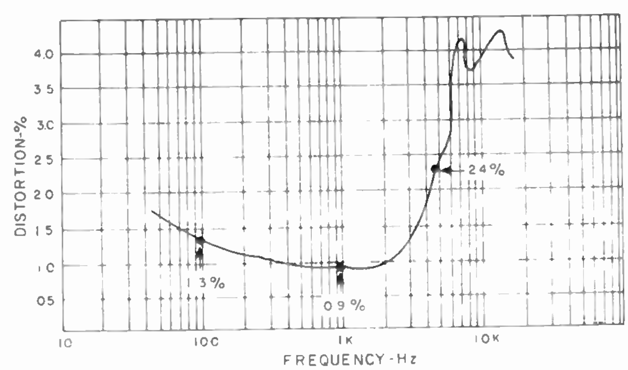


Fig. 6—THD in stereo versus frequency will show up undesirable "Beat" signals.

How to evaluate FM stereo tuner performance

Daniel R. von Recklinghausen*

EVERY YEAR buyers' guides are published which present a summary of the performance characteristics of tuners and receivers. Every month magazines publish reviews of this equipment in which they present the findings of their tests. Seemingly every week advertisements, in newspapers or magazines, in catalogs or through the mail, vie for the prospective buyer's attention. Once the prospect is ready to buy a tuner or receiver, he may receive advice from his friendly salesman in addition to the advice he has gotten from his friends, neighbors, and relatives. How can he assimilate all this and then make a rational decision?

There should be several factors influencing a decision. Broadly, they may be classified as convenience factors (or "features"), cost, and performance capabilities.

The most important convenience factor which may influence a decision to select a tuner or receiver is the desire to have all operating controls on one piece of equipment. In this case a receiver (or tuner-amplifier combination) is the obvious choice. If little space is available to install the electronics, the same choice should be made.

Tuners, without the added audio controls of an amplifier, generally permit greater flexibility in connecting a high fidelity music system and permit equipment of various manufacturers to be used. This simplifies the up-grading or modification of such a system at a later time.

Tuners often duplicate some controls of an amplifier, such as noise filters or level controls. They also require their own chassis and enclosure with panel and knobs, as well as power supply components to operate the receiving circuits. Since many of these parts are also required in an amplifier, this duplication makes the combination of a separate tuner and a separate amplifier more costly than a receiver. In the early days of high fidelity equipment, when only tube circuits existed, there was the feeling that the tuner portion might be of a lower quality in receivers than in tuners. Principally, the heat generated by the vacuum tubes was thought to cause the tuner circuits to drift. Even back in the "tube" days, the designers were able to lick this problem, which now, in the "solid state" days, is no longer of any consequence. Tuner performance in receivers can be and is fully as good as found in separate tuners.

The last, hardest and the most important step remains: The evaluation of the performance capabilities of a tuner. Only one question really need be answered: How many stations' programs will the tuner reproduce and how well? It is often impossible to compare all tuners under consideration side-by-side in one's home, comparing the quality of each station's signal, because for a complete listening test, signal quality and interference susceptibility must be compared for each tuning position—a truly imposing task.

The majority of high fidelity equipment users live either in the city or its suburbs. The majority of the 4350 AM stations in the United States, the majority of the 2750 FM stations, and the majority of the 900 TV stations are located in metropolitan areas. The local AM and FM stations will most likely carry different programs since the Federal Communications Commission (FCC) ruled that AM and FM stations under the same ownership may not duplicate more the 50% of their programs if they are assigned to the same city of more than 100,000 population. Unless all the AM programs are of no interest to the listener, or he has enough AM radios, it is wise to consider the purchase of an AM/FM receiver or tuner rather than the FM-only variety.

Even though AM broadcasting stations have to provide at least 5000 Hz frequency response, and many of them provide as much as 15,000 Hz response, interference from other stations and many electrical appliances will make wide frequency range listening difficult in most locations. For this reason the FM reception capabilities of a receiver are considered more important.

Again, keeping the metropolitan FM listener in mind, the FM tuner will have on the average three or four strong local signals (and perhaps as many as 24) picked up by its antenna. These signals will tend to intermodulate with each other in the RF amplifier and converter stages of the tuner, causing spurious signals and crossmodulation products to be generated. These stray signals will usually appear on frequencies other than those occupied by the local signals. They may masquerade as "other" stations but really are nothing but a repetition of the local signals. If these signals were to appear only on unoccupied channels they would cause no harm except that of repetition. More than likely, these stray signals will interfere with weaker, more distant signals and may even obliterate them. Consequently, the number of listenable FM stations is reduced from what it could be.

In choosing a tuner it is wise to select one which resists overload due to strong signals, i.e., a tuner which has good crossmodulation rejection. Crossmodulation rejection (or spurious response rejection) is measured in dB in accordance with the standard of tuner measurement published by the Institute of High Fidelity. The number may be thought of as representing the "dynamic range" of r.f. signals the tuner is capable of accepting without problems.

At least 70 dB crossmodulation rejection should be available for metropolitan FM reception if other than local stations only are considered as program sources. Higher numbers are, of course better, with each additional 6 dB permitting signals of twice the field strength (corresponding to approximately one-half the distance to a transmitting antenna) to be available to the tuner without trouble.

Since the IHF crossmodulation measurement is performed with only one interfering signal, and since interference may be caused by several signals of varying strengths, a high cross-

*Technical Director, H. H. Scott, Inc.

modulation figure is not an absolute guarantee against all forms of tuner overload due to strong signals but it is still the best available performance indicator. A set of interference measurements on one tuner under all possible signal conditions would take a very long time. A full report might be the size of one volume of an encyclopedia and would be harder to interpret.

An abundance of strong and weak FM signals are the typical conditions in a metropolitan area. In the writer's home (in a suburb of Boston) 41 different listenable FM signals were recently counted using a high-quality tuner operating from a folded dipole antenna located indoors on the ground floor of a frame building.

To receive this many FM stations (when available) requires a tuner of high selectivity. FM stations are assigned by the FCC to 100 channels 200 kHz apart, ranging from 88.1 to 107.9 MHz. In any one locality, stations are assigned to channels 800 kHz or more apart. Stations in nearby areas are assigned to frequencies in between. These may be assigned to the adjacent channel (200 kHz away), the alternate channel (400 kHz away) or perhaps the third channel (600 kHz away).

Selectivity for an FM tuner is measured for the interference experienced when tuned to the desired channel, with a stronger interfering signal fully modulating the alternate channel. It is measured in dB, with a higher number indicating better performance.

Selectivity and bandwidth of a tuner are linked together. While selectivity may be thought of a measure of rejection of signals away (in frequency) from the desired signal, bandwidth may be thought of as denoting the width of the signal spectrum of the desired station accepted by the tuner.

The width of the signal spectrum depends not only on the maximum deviation of the instaneous station frequency from its average (and assigned) carrier frequency, but also on the modulation frequency. The FCC defines 75 kHz deviation as 100% modulation for an FM broadcasting station. Since audio signals have both plus and minus values the station frequency varies both above and below its average value, resulting in instantaneous frequencies spread over a range of 150 kHz. The actual spectrum occupancy of the transmitted signal is larger because the actual transmitted frequencies are not instantaneous but involve an infinite number of "sidebands" (of the carrier) when the carrier is deviated at a modulation frequency rate. These sidebands decrease in strength at frequencies further away than the "deviation" from the carrier, yet the tuner must process all of them for truly distortion-free reception. This could not permit any selectivity because of the requirements for "infinite" bandwidth. The elimination or attenuation of sidebands away from the carrier causes distortion. Calculations for a special ideal (and therefore mathematically treatable) filter indicate bandwidths of 265 kHz and 225 kHz if harmonic distortion of 1% and 3% respectively may be tolerated at certain modulation frequencies. Practical, well-designed filters produce less distortion than these figures for the same bandwidths. Since such a bandwidth includes some of the channel space assigned to adjacent-channel stations, and since selectivity is measured with a 100% modulated signal, it is evident that very little selectivity is obtained for adjacent channel signals. The figure may even be negative, indicating the need for a stronger desired signal compared to the interfering signal. For this reason, selectivity is measured for alternate channel signals.

The full story of selectivity is not found in a single selectivity number, but in a selectivity curve as seen in Fig. 1. Here, the desired signal is indicated by a cross, denoting its strength and relative frequency. The interfering signal is represented by a curve. This curve shows the frequency of the interfering signal and its strength adjusted so that the desired signal predominates in the audio output and therefore "captures" the interference. This selectivity curve (made according to the procedure outlined in the IHF tuner standard) is a dynamic selectivity curve and does not show the response of a selective filter measured in the conventional manner.

At a frequency difference of 0 kHz a special selectivity point is measured—it is the capture ratio. This ratio indicates the ratio of signals required for the tuner to reproduce the stronger of the pair while rejecting the weaker.

In the example shown, the selectivity curves are shown for a desired signal equal to that giving IHF usable sensitivity and for a desired signal 20 dB stronger. For weak signals, the residual noise provides a second interfering signal, resulting in the slightly wiggly curve shape between ± 100 kHz. At stronger signals, the noise influence is eliminated and the true two-signal selectivity curve emerges. The measured "capture ratio" is also improved.

From these curves, it can also be seen that weak adjacent channel signals may be received if the receiver is tuned off to the side so as to discriminate against the interfering signal. Of course, detuning causes distortion, but detuning is usually made so as to minimize the total of audible distortion and interference products.

Selectivity, when reported as a single number, denotes the point on the selectivity curve which is 400 kHz from the desired (center) frequency. For asymmetrical selectivity curves, the center frequency is assumed to be located halfway between the two (low and high frequency) intersections of identical selectivity spaced by a total of 800 kHz.

Selectivity curves of practical tuners are always slightly asymmetrical because coupling of the selective elements is generally by some reactive rather than a resonant elements. Selectivity curves of tuners which use mechanical elements, such as quartz crystal or ceramic resonators, as selective elements usually show substantially reduced selectivity at the higher frequency portion of the selectivity curve and also show a substantial reduction of the slope at selectivities in excess of, say, 45 dB. This is usually caused by a combination of electrical and mechanical coupling and by other than desired resonant modes of the ceramic or crystal elements.

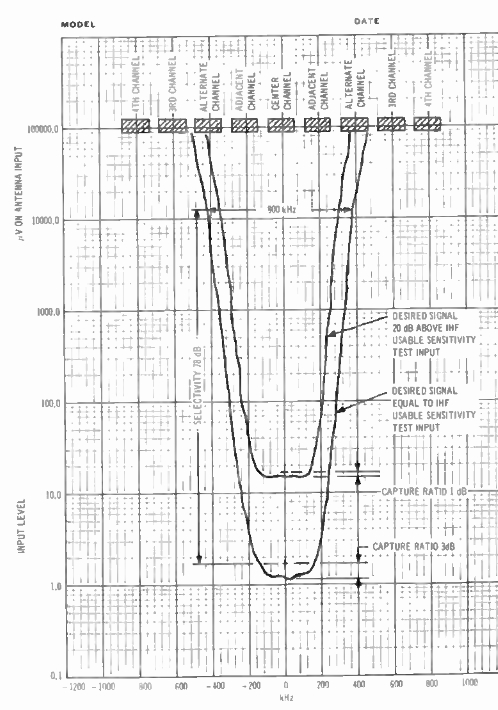


Fig. 1—Selectivity curves and level of desired signal.

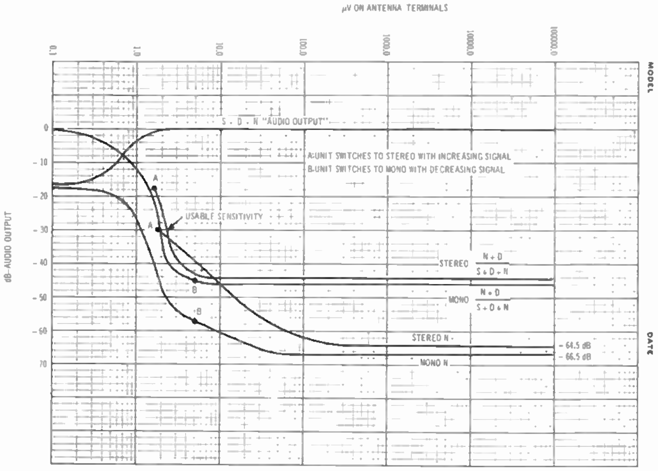


Fig. 2—FM performance.

Overload of the input circuits of a tuner will also cause this widening of the selectivity curve.

The next set of curves which should be examined when selecting a tuner are those often published under the general title of FM characteristics. An example of these is shown in Fig. 2. The first curve of interest is the "audio output" curve, showing total tuner output voltage due to 400 Hz, 75 kHz deviation monophonic modulation with respect to the (300 ohm) antenna input signal. This output contains the audio signal, its distortion, and whatever noise there may be; it is labelled "S + D + N."

The ideal tuner shows this output to be constant for input signals in excess of approximately 5 μV. At the signal level corresponding to IHF usable sensitivity, the audio output will have decreased by approximately 1 dB and in the absence of any input signal will be approximately 15 dB lower than its "normal" output value. These figures assume that the limiter is fully limiting with the random noise generated in the first amplifying stage of the tuner. Only by this action can one be assured that the tuner is sensitive to frequency modulation only and does not respond to amplitude modulation.

At this point it should be noted that the term "full limiting" has been misused—often denoting some arbitrary relative audio output or signal-to-noise ratio rather than its proper meaning—the removal of amplitude modulation.

Practical FM tuners approach the ideal S + D + N curve very closely. Tuners which show markedly lower (by more than 6 dB) audio output at 0 input signal probably have inadequate gain for weak signal processing and may not be fully adequate for critical listening. Tuners which show a variation in audio output of more than 2 dB for signals stronger than their usable sensitivity signal, may be inadequate for many high-fidelity installations because the normal audio output may vary audibly during moderate signal fading experienced when aircraft cause the received signal to vary (a phenomenon known as "airplane flutter") by adding a varying multipath reflection to the direct station signal.

The second set of curves, labelled "mono N" and "stereo N" show the residual noise output of the tuner when subjected to an unmodulated ("mono N") carrier signal or one modulated with a 19 kHz stereophonic pilot signal ("stereo N").

At moderate to high input signals, these curves will be horizontal, showing the total residual noise of the tuner and generator circuits. The "stereo N" curve may also contain residual amounts of the 19 kHz and 38 kHz signals passing through the audio filters of the tuner and some stereo modulator and demodulator noise.

The distance in dB of the "mono N" curve to the S + D + N curve shows the monophonic signal-to-noise ratio of the tuner, and the distance of the "stereo N" curve to the S + D + N curve shows the stereo signal-to-noise ratio. The rated values are usually found at the 1000μV level.

It should be noted that the "N" curves merge with the "S + D + N" curve for a 0 input signal.

In the range of the IHF sensitivity input, the "mono N" curve for the ideal tuner shows its steepest decent, with the noise output decreasing in the steepest portion as much as 6 dB for a 1 dB increase in signal level. Non-ideal tuners will show a lesser figure.

Once the mono signal-to-noise ratio has reached about 48 dB, the noise level will decrease proportionally to the signal increase (a 1 dB signal increase causes a 1 dB noise decrease) until the ultimate signal-to-noise ratio has been reached. The 48 dB point shows the "knee" for mono performance, while the 25 dB point of the stereo signal-to-noise ratio is the "knee" of the stereo performance.

The third set of curves show the total relative undesired output of the tuner, i.e., the total of noise and distortion with respect to the total output. They are labelled "mono N + D/S + D + N" and "stereo N + D/S + D + N". In the absence of any signal, these curves approach 0 dB, and at moderate to high inputs they show the total distortion of the tuner. At all levels, these curves have a higher value than the corresponding "mono N/S + D + N" and "stereo N/S + D + N" curves (which are not shown). In the steep transition region, the corresponding mono or stereo curves should be no more than a few dB apart. Otherwise, the tuner may be suspected to be of narrow bandwidth—which should also be reflected by relatively high stereo distortion.

The signal level at which the "mono N + D/S + D + N" curve crosses the -30 dB line is the point of IHF usable sensitivity—and the rated value for this is the highest number of microvolts found at the carrier frequencies of 90, 98, and 106 MHz.

As may be imagined, listening to programs at the usable sensitivity input is not a hi-fi listening experience, but measurement at this level can be repeatedly performed and is valuable as a basis for comparison. For example, enjoyable mono listening requires signal-to-noise ratios in excess of 50 dB. Some tuners may reach this at the IHF input level, and others may require as much as 6 dB higher signal level. Here, the shape of the curves will help decide this fact.

Enjoyable stereo listening also requires at least 50 dB signal-to-noise ratio. As can be seen, substantially higher signal levels are required—from 10 dB to over 20 dB above the IHF test input.

These figures illustrate that substantially more useful information can be obtained from a few sets of curves than from a list of numbers.

How can a metropolitan listener evaluate the various sensitivity curves? If he lives in the middle of the city in a tall steel-and-concrete building without access to an outdoor antenna he faces most difficult signal reception conditions. Not only will he not be likely to be able to receive the direct unreflected signal, but he will be subject to a conglomerate of signal reflections, varying considerably in signal strength from fairly weak to very strong. His signals will be often pure "multipath" and laced with pulse noise interference from automotive traffic and other sources. Such a listener should choose his tuner to have good spurious signal (crossmodulation) rejection and to have good performance for signals of weak to high intensity. Extreme sensitivity is not required because signals below 5μV may be riddled with man-made noise. The selectivity requirements in the city are usually moderate (30 dB or more), because distant signals will be severly attenuated and often buried in noise.

The suburban listener usually has better signals available; generally all of them, local and distant, are stronger than those "downtown." His tuner still must have excellent spurious response rejection. Since listenable distant signals are available, a prime requirement should be good selectivity (40 dB or more). Sensitivity of a tuner need not be extreme, but must not be sacrificed at the expense of spurious responses. Since the man-made noise is lower than in the city, good signal-to-noise ratio with moderate to strong signals should be sought.

The minority of rural listeners should place sensitivity of a tuner high on the list, followed by good signal-to-noise ratio. Most important is a good installation of a high-gain directional antenna to capture as much as possible of the little signal available. An antenna rotator is a "must."

Two very important FM tuner characteristics are usually not reported in curve form and one of them is even difficult to express as a number. The first of these is the AM suppression.

The signals broadcast by the FM station are modulated by varying the transmitter's frequency while keeping its power output constant. All the tuner circuits in the transmitter and its antenna system are usually adjusted to minimize any output variation. The tuner circuits in the receiver usually are of narrower bandwidth than those of the transmitter since they have to select the desired signal and reject signals on other frequencies.

For this reason, the received signal at the input to the limiter stages will be attenuated by 0.5 to 3 dB (in typical tuners) as the signal varies from its center frequency to the maximum deviation. This in effect is incidental amplitude modulation of the FM signal, occurring at twice the audio modulation frequency. If the tuner were fully sensitive to this amplitude modulation, its detected signal would experience 3% to 17% distortion. Fortunately, the limiter circuits (along with the detector circuit) remove most of the amplitude modulation, and low distortion reception is possible. The effectiveness with which these circuits remove this amplitude modulation is known (and rated) as amplitude modulation rejection and is measured in dB. Any high fidelity tuner worthy of the name should have at least 40 dB AM rejection when measured with 30% AM. Claims of AM rejection in excess of 60 dB are within the realm of the possible, but are difficult to verify since normal laboratory test equipment is generally not capable of generating AM signals sufficiently free of incidental frequency modulation.

Good AM rejection in a tuner is required not only for removal of incidental amplitude modulation generated within the selective circuits but also for removing as much audible distortion as possible when receiving signals under multipath signal conditions.

As the name indicates, multipath means that the signal reaches the receiver's antenna by a multitude of paths, each having a different loss, and because of their length, a different delay. The effect may be demonstrated as shown in Fig. 3a. Here, just two signal paths are considered, a direct path and a path involving a reflecting obstacle (a hill or a group of tall buildings) located about one mile away. This would cause the reflected signal to have a delay of approximately 10 microseconds. The direct and reflected signals add vectorially in amplitude and phase as shown in Fig. 3b. The resultant now has neither the amplitude nor the phase of the original signal. As the frequency of the signal changes, the relative phase of reflection to direct signal changes 360° for every 100 kHz of signal change. Thus, there may be as many as two instantaneous signal maxima or minima within the normal ±75 kHz modulation range in this example. The resultant amplitude modulation may now include strong components of fourth harmonic distortion of a strength determined by the ratio of direct-to-reflected signal.

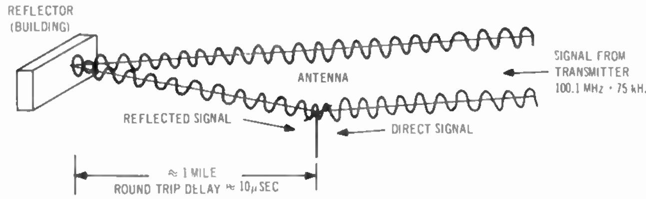


Fig. 3a—Simple multipath caused by one reflected signal and one direct signal.

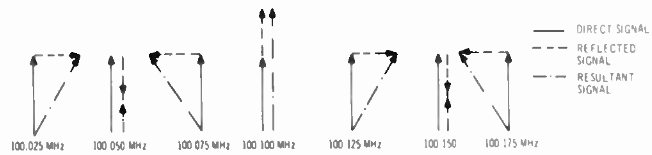


Fig. 3b—Vectorial addition of direct and reflected signals vs. frequency.

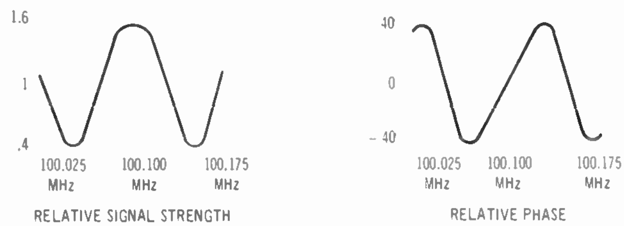


Fig. 3c—Amplitude and phase response of total transmission path containing one reflection.

The limiter circuits will probably remove most of the incidental amplitude modulation, but they will not affect the phase of the resultant. This phase now varies with deviation of the carrier. Since the carrier is deviated by the modulation signals, the resultant also phase modulates the signal.

This in turn is an incidental frequency modulation of the signal—but one increasing with modulation frequency and containing practically pure distortion. Consequently, under multipath reception conditions, distortion increases with modulation frequency and is most serious at the highest modulation frequencies—those which contain the stereo sub-channel. This is the main reason for stereo reception in cities being often unacceptable when monophonic signals sound much better.

No high fidelity tuner can remove multipath signals. (Extremely complex circuits may be able ot minimize multipath problems, but they require laborious adjustment, and so far only the military services have been able to experiment with them). The best cure for multipath problems is the use of a directional antenna along with careful orientation and adjustment. Even indoor antennas of the "rabbit ear" kind can be used with good effect and may be most practical when readjustment is required for each signal.

To minimize multipath problems requires one additional tool, the multipath indicator. As shown above, the increased high-frequency distortion or stereo distortion may be used with the ear acting as an indicator. Unfortunately, the ear is not a reliable tool, particularly when the broadcast station does not "cooperate" by playing music containing a lot of high frequencies or by playing stereo material.

The best results can be obtained when a multipath indicator is connected to the tuner. An oscilloscope may be connected to many tuners to display multipath. Essentially, these connections involve vertical trace deflection with an output containing a signal proportional to signal strength, and horizontal deflection with the FM detector output. Detailed information for connections will have to be obtained from the manufacturers of these tuners, or their representatives or dealers.

Other tuners may have these connections already provided and some of them contain built-in multipath indicators such as oscilloscopes or meters. By just being able to adjust the antenna for the best signal, these tuners will be able to give results superior to other installations when difficult multipath reception conditions exist. These difficult conditions can be predicted with good accuracy in most any city having tall buildings, and in any hilly or mountainous area. Flat country or over-water signal paths are rarely subject to serious multipath problems.

The next important tuner characteristic not reported as a curve (or even a number) is the tuner's pulse noise rejection. As shown above in the section on selectivity, the capture ratio indicates the tuner's ability to have the stronger of two signals predominate in its output while suppressing the weaker one.

When a tuner is subjected to a noise pulse, the pulse should "capture" the signal from the station for the length of time in which the pulse is stronger than the signal. The strength of the pulse in the FM band caused by a passenger automobile ignition system may be 1000 microvolts per meter at a distance of 500 feet. This is 20 times stronger than the old "grade B" service contour of 50 microvolts per meter of an FM broadcast station.

It is important that the pulse cause no worse problems than capture of the signal. As soon as the pulse becomes weaker than the signal, the signal should predominate and no transient due to previous pulse overload should remain. A single pulse should sound as a "tick" pretty much regardless of its strength and not as a "pop". In particular, a pulse or a series of pulses, should not be audible with the receiver tuned off-station. If pulse noise were audible under this condition, listening quality for other than local stations may be impaired.

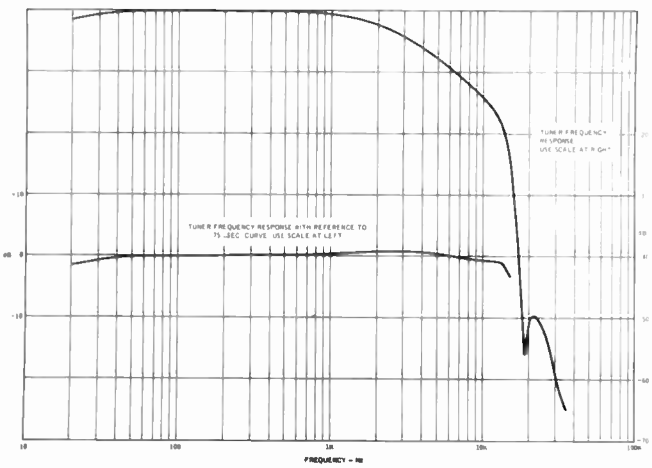


Fig. 4—Tuner frequency response.

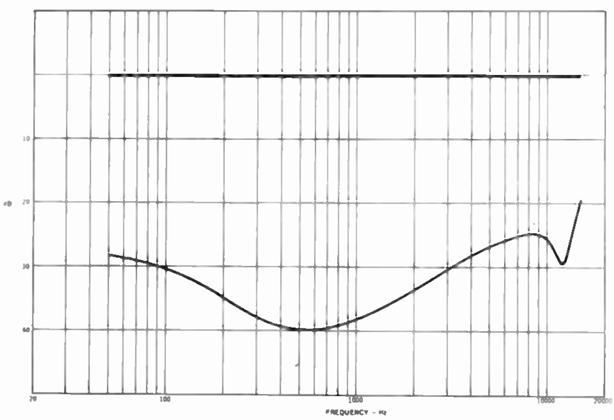


Fig. 5—FM stereo separation.

All the previous tests and curves have documented the important aspects of tuner performance from antenna to the final FM detection. The group of curves in Figs. 4 and 5 describe the performance of the circuits which handle the detector output.

Figure 4 shows the audio frequency response of the tuner circuits. It is shown as a normal frequency response, including the response above 15,000 Hz, the maximum audio frequency broadcast by a stereo FM station. The region above 15 kHz shows how well the 19 kHz pilot signal and the 38 kHz re-insert carrier signal are removed from the tuner output. These frequencies may cause intermodulation with the bias oscillator in tape recorders and the recording of whistle frequencies.

The frequency response is also shown with respect to the standard 75 microsecond de-emphasis curve. It shows how accurately the signals are reproduced. This error should be less than ± 2 dB between 50 Hz and 15 kHz, the limit frequencies for FM broadcast stations.

Figure 5 shows the stereo separation of a tuner. As may be appreciated, more information can be gleaned from a curve than from a single figure, say, 30 dB separation and without specification of frequency.

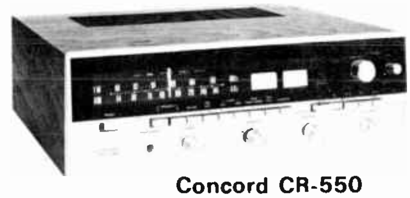
For good listening, the tests performed by the Bell Laboratories and by General Electric show that at least 20 dB separation be available over the range of 100 to 8000 Hz. Most tuners are capable of exceeding this figure-but only a curve will prove it.

There are many more tests which can be and are performed on tuners. They may deal with more refined aspects of the characteristics discussed above, or they may deal with squelch performance, tuning indication, drift, or many other factors. In all cases, the basic characteristics analyzed here should be considered most when choosing a tuner or receiver. Only after these have been evaluated with respect to the expected signal conditions for city, suburb, or country should other convenience features not discussed here be analyzed.

Last, but not least, reliability and reputation, leadership in technology, and soundness of design should be evaluated. The tuner bought today should give years of good service. Æ

Receivers

Altec 710



Concord CR-550

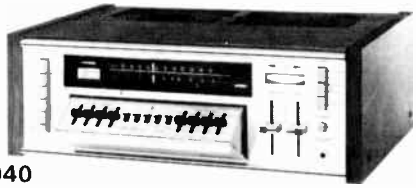
Akai AS-980

Notes: (1) All models solid-state except when model number is preceded by (T).

(2) "K" indicates kit price; "W" indicates wired price.

	AMPLIFIER										TUNER													
MANUFACTURER	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 ±? dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	IHF sensitivity, µV	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %	THD stereo, 100% mod., %	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	SPECIAL FEATURES
AKAI	AS-980 4-chan.	40	0.03	1.0		10-60k	5-80k −3	85	3		2	2		0.2	0.6	40	Mtr.	80	6	Yes	26 x 17 x 6¾	45	799.95	SQ, matrix, CD-4; rem. contl. avail.; 4 input VU mtrs. FM 4 ch. output; rosewood cab.
	AS-970 4-chan.	30	0.03	1.0		10-60k	5-80k −3	85	3		2.0	2		0.4	0.8	40	Mtr.	70	4	Yes	23 x 15½ x 6¾	36	599.95	SQ, matrix, opt. front pnl. rem. contl.; l. & r. chan. VU mtrs.; FM 4-chan. output, rosewood cab.
	AS-960 4-chan.	25	0.03	1.0		10-60k	5-80k −3	85	3		2.3	3.5		0.4	0.8	35	Mtr.	70	1	Yes	19 x 16 x 6¾	28	499.95	SQ, matrix; FM 4-chan. output, rosewood cab.
	AA-910	18	0.2	1.0		20-45k	20-70k −3	75	3		2	2.5		−0.4	−0.8	40	Mtr.	65	1	Yes	18 x 14½ x 6½	24	229.95	Vol. contl. w/mem. set.; front pnl. stereo mic inputs; FM sens. swit. Model AA-910B; built-in Dolby & 3 mtrs.
	AA-930	37½	0.03	1.0		10-60k	7-80k −3	85	3		1.8	1.5		0.2	0.6	40	Mtr.	80	2	Yes	19 x 15½ x 6½	33	399.95	May be used as sep. amps & preamps.
ALTEC	704	12½	0.5	0.5	0.5	10-30k		60	2.5		2.5	3	20-15k ±1	0.5		35	Mtr.		1	Yes	15¾ x 13½ x 4¾	15½	199.95	
	710	30	0.5	0.5	0.5	10-30k		60	2.5/8		2.5	1.5	20-15k ±1	0.5		40	Mtr.		1	Yes	16½ x 15½ x 5⅝	23½	299.00	
	714	44	0.5	0.5	0.5	15-25k		60	2/5		1.9	2	20-15k ±1	0.5		40	Mtr.		2	Yes	16½ x 13⅜ x 5⅝	34	399.00	
	725	60	0.3	0.3	0.3	20-20k ±½		60	2/5		1.8	1.3	20-15k ±1	0.3		40	Mtr.		2	Yes	17¾ x 16½ x 5	48	599.00	
AUDIOSON-KIRKSAETER	RTX 120.85	85	0.09			8-30k	6-50k −3	90	1.5		1.5	2.5	10-15k	0.2	0.8	36		60	2	No	19 x 14 x 5¼	31	700.00	
	RTX 85.55	65	0.09			8-30k	6-50k −3	90	1.5		1.5	2.5	10-15k	0.2	0.8	36		60	2	No	19 x 14 x 5¼	29	600.00	
B & O	4000	60	0.1	0.3	0.3	10-35k	20-30k ±1.5	65	3		2	3	20-15k	0.4	0.4	35	Mtr. & Lt.	55	1	No	22¾ x 10¾ x 3¾	22	520.00	Var. diode tuning w/preset sel. of 6 stations.
	3000-2	40	0.5	0.6	0.6	20-30k	30-30k ±1.5	65	3		2	3	20-15k	0.4	0.4	35	Mtr. & Lt.	55	1	No	22¾ x 10¼ x 3¾	19	400.00	Same as above.
CONCORD (BENJAMIN)	CR100	6	1.0			27-30k	28-25k ±1	58	2.8		3.5	2	20-15k	1.0	1.5	30	Mtr.	40	1	Yes	16 x 11½ x 4¼	10	129.95	
	CR200	12	1.0			26-33k	23-38k ±1	58	2.8		2.9	1.8	20-15k	1.0	1.5	35	Mtr.	43	1	Yes	15⅜ x 12¾ x 5	16	179.95	Fine tuning; duo-glow ind.; elect. ckt. breaker.
	CR250	25	1.0			20-35k	22-40k ±1	60	2.5		2.3	1.5	20-15k	1.0	1.5	35	Mtr.	46	2	Yes	18¾ x 14 x 5¼	20	229.95	Same as above.
	CR550	55	0.5			20-60k	10-40k ±1	65	2.2		1.7	1.5	20-15k	0.7	1.2	40	Mtr.	55	2	Yes	17⅝ x 15½ x 6	31	379.95	Fine tuning; Glow-point™—tuning; pushbutton contl. swits.; ckt. brkr.

Receivers



Dokorder MR-940

Fisher 504

Notes: (1) All models solid-state except when model number is preceded by (T).

(2) "K" indicates kit price; "W" indicates wired price.

MANUFACTURER	MODEL	AMPLIFIER: 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 ±? dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	TUNER: IHF sensitivity, µV	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %	THD stereo, 100% mod., %	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	SPECIAL FEATURES
DOKORDER	MR-900	100	0.3	3.0		30-45k	30-40k	45	2.5; 150		1.5	1.5		0.5	0.5	35	Mtr.	50	1	Yes	20 x 14 x 6¾	33	439.95	"Kangaroo pouch" contl. pnl.; FET frnt.-end; FM mut.; tape mon.; term. for 2 sets spkrs., 2 phonos, 2 aux, tape.
	MR-940 4-chan.	30*	0.3	3.0		30-45k	30-40k	45	2.5; 150		1.5	1.5		0.5	0.5	35	Mtr.	50	1	Yes	20 x 14 x 6¾	33	539.95	Matrix & discr. 4-ch. E-V decod.; BTL bridging; fold-away contl. pnl. 4-ch. tone, vol. bal. contls.; *4-chan., 58W./chan. in 2-ch. mode.
FISHER	180	36	1.0	0.5	0.1	30-20k	20-20k ±2	60	2.5	50	2.5	3	20-15k	0.5	0.8	35	Mtr.	40	1	Yes	16½ x 12 x 5¾	19	249.95	Model 170 similar, but 28W./chan. & pwr. bndwdth. 40-20k. $199.95.
	4060 4-chan.	60	1.0	0.8	0.3	30-30k	20-20k ±2	60	2.5	45	2.5	2.5	20-15k	0.5	0.8	35	Mtr.	40	2	Yes	19¼ x 16¾ x 5⅝	24½	369.95	Strpd. amps.; 4020 sim. but 40W./chan. & 1 meter, $299.95.
	304B 4-chan.	80	0.5	0.8	0.15	12-30k	20-20k ±2	56	2.7	60	1.8	1.2	20-15k	0.2	0.3	38	Mtr.	60	1	Yes	21½ x 16⅞ x 6⅞	39	399.95	Joystick bal. contl.; 304 same but w/o joystick, $329.95.
	504 4-chan.	160	0.5	0.8	0.15	8-40k	20-20k ±2	56	2.7	60	1.8	1.2	20-15k	0.2	0.3	38	Mtr.	60	2	Yes	21½ x 16⅞ x 6⅞	43	599.95	404 sim. but 112W./chan., $429.95.
GTE SYLVANIA	CR2742A	25	0.5	0.5	0.2	17-35k	17-35k ±1.5	50	2.6	80	1.9	1.5	20-15k	1.0	0.6	40	Mtr.	55	1	Yes	16⅞ x13 x 5⅜	24	219.95	Sylvania Phase Q-4 matrix.
	CR2743A	50	0.5	0.5	0.2	17-35k	17-35k ±1.5	50	2.6	80	1.9	1.5	20-15k	1.0	0.6	40	Mtr.	55	1	Yes	16⅞ x 15½ x 5⅜	30	299.95	As above.
	RQ3747 4-chan.	25	0.5	0.5	0.2	18-35k	20-30k ±1.5	60	2.6	100	1.9	1.5	20-15k	1.0	0.6	35	2 Mtrs.	55	2	Yes	21¼ x 15 x 6⅞	35	449.95	Brdg. strp. ckt. gives 60W/60W RMS in 2-ch.
	RQ3748 4-chan.	50	0.5	0.5	0.2	18-35k	20-30k ±1.5	60	2.6	100	1.9	1.5	20-15k	1.0	0.6	35	2 Mtrs.	55	2	Yes	21¼ x 15 x 6⅞	38	549.95	Brdg. strp. ckt. gives 125W/125W RMS in 2-ch.
GROMMES (PRECISION)	600	20	0.3	0.5	0.1	20-20k	20-20k ±1	60	2	75	3.5	5	20-15k	0.5	0.7	30	Mtr.	40	1	Yes	14 x 4¾ x 6½	17	179.95	
	503A	40	0.3	0.5	0.1	20-20k	20-20k ±1	60	2	75	2	2.5	20-15k	0.5	0.7	35	Mtr.	45	1	Yes	16 x 5¼ x 13	30	359.50	
HARMAN KARDON	900+ 4-chan.	32*	0.5	0.15	0.15	10-40k	1-100k ±1	65	3	85	1.8	1.6	50-15k	0.4	0.5	37	Mtr.	60	1*	Yes	20⅛ x 17 x 6¼	45	749.95	CD-4; SQ; "quieting" mtr.; 4-ch. FM output; *2-ch., 90W/chan.
	800+ 4-chan.	22*	0.5	0.15	0.15	10-40k	1-100k ±1	65	3	75	2.0	2.5	50-15k	0.6	0.7	35	Mtr.	40	2	Yes	18⅛ x 16¼ x 6¼	39	599.95	As above, *2 ch. output 50W/chan.
	700+ 4-chan.	18*	0.5	0.15	0.15	10-40k	5-70k ±1	65	3	60	3	2.5	50-15k	0.6	0.7	35	Mtr.	40	1	Yes	18⅛ x 16¼ x 6¼	34	499.95	As above, *4-ch. output.
HEATH	AR-14	10*	1.0*	1.0		7-90k	12-60k ±1	60	4.5		5	3.0	20-15k	1.0		30				No	15¼ x 12 x 3⅞	14	99.95K	*Full pwr. 20-20k at rated THD (both chans. driven); stereo 'phone jack; spkr. swit.
	AR-1214	15*	0.5*	0.5	0.2	5-30k	7-100k ±1	65	2.0	75	2	2.0	20-15k	0.5	1.0	40		60		Yes	17 x 13 x 3⅞	13	169.95K	*Pwr. rating as above; cer. filts.; PLL mpx; AM.
	AR-1302	20*	0.25*	0.25	0.1	5-30k	6-35k ±1	65	2.4	155	1.9	1.8	20-15k	0.5	0.5	40	Mtr.	60	2	Yes	16¾ x 14½ x 5⅛	26½	239.95K	*Pwr. rating as above; mod. constr.; cer. filts.; preassem. FM front end; 2 spkr. swit.
	AR-29	35*	0.25*	0.2	0.1	5-30k	7-60k ±1	65	2.2	155	1.8	1.5	20-15k	0.5	0.5	40	Mtr.	70	2	Yes	16¾ x 14½ x 5⅛	26½	299.95K	*Pwr. rating as above; mod. constr.; 9-pole L-C filt.; pre-assem. FM front end; 2 spkr. swit.
	AR-1500	60*	0.25*	0.1	0.1	8-30k	7-80k ±1	63	1.8	145	1.8	1.5	20-15k	0.5	0.5	40	Mtr.	90	2	Yes	18½ x 13⅞ x 5⅛	32	379.95K	*Pwr. rating as above; mod. constr.; LC filts. in FM/AM; noise oper. squelch; tape mon.; tone flat.

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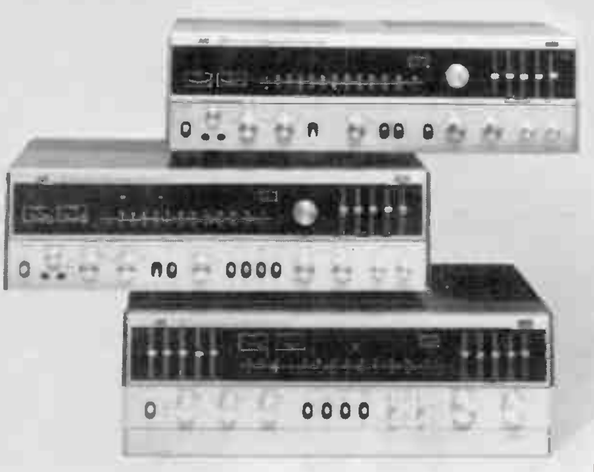

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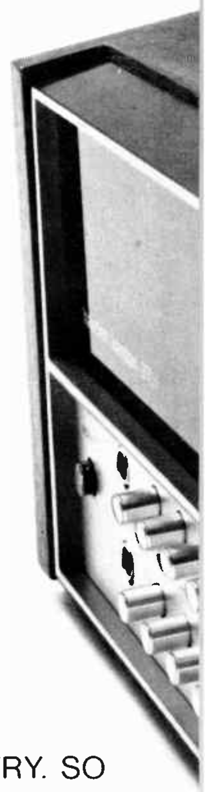

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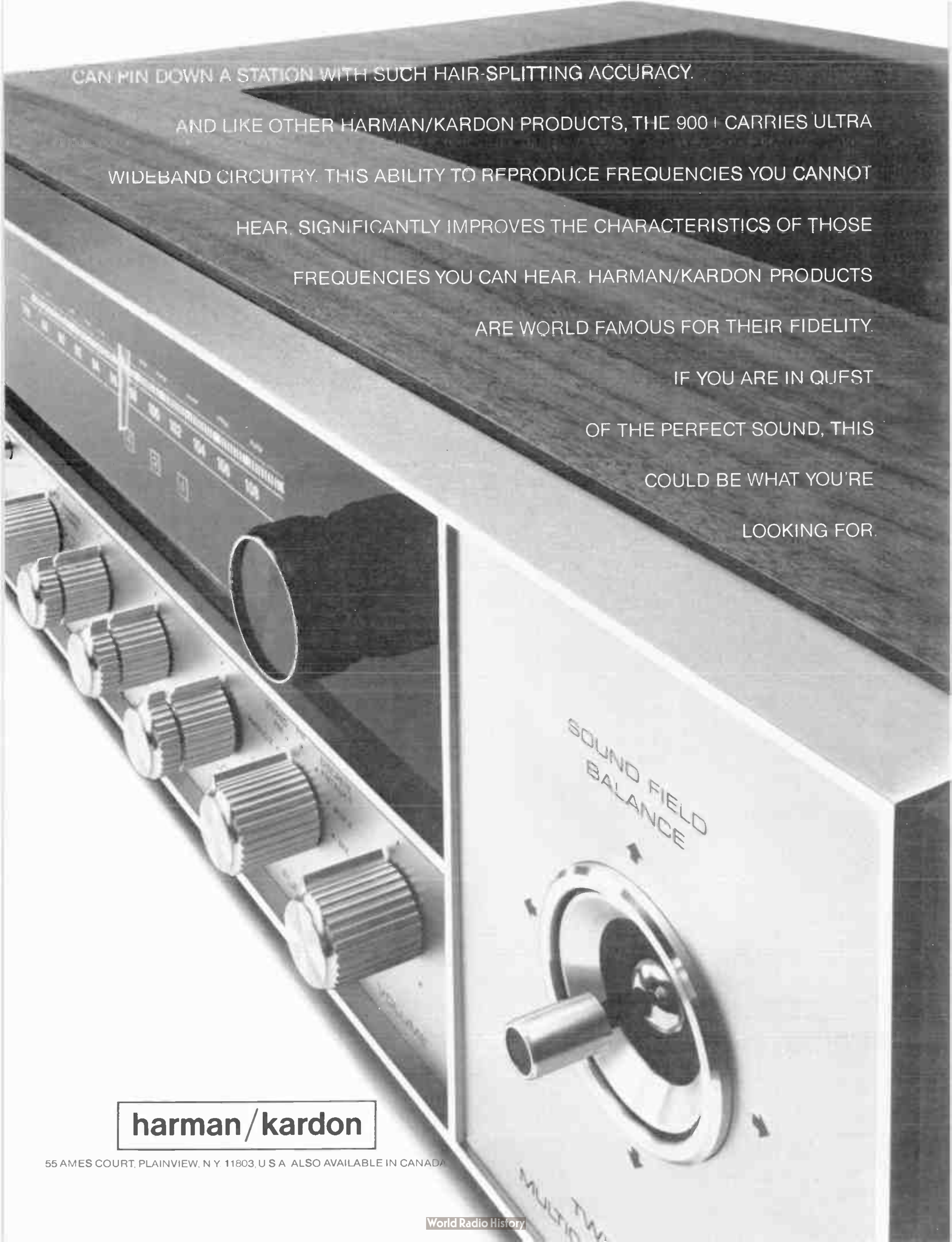
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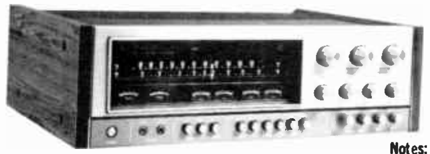
Receivers

JVC 4VR-5456



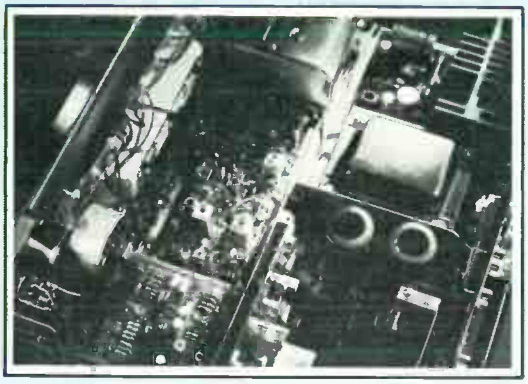
Kenwood KR-9340

Notes: (1) All models solid-state except when model number is preceded by (T).

(2) "K" indicates kit price; "W" indicates wired price.

MANUFACTURER	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 ±? dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	IHF sensitivity, µV	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %	THD stereo, 100% mod., %	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	SPECIAL FEATURES
		AMPLIFIER									TUNER													
HERVIC	H150	75	0.25	0.25		5-100k	20-20k ±0.5	72	25		1.7	1.8	20-15k	0.15		50	Mtr.	110	2	No	22 x 14 x 8	50	779.95	
HITACHI	SR-3200	13	1.0			25-40k +0 −3		70	2.5/ 5.0		2.5	1.2			0.8	36	Mtr.	46	2	Yes	17 x 5 x 15	18	229.95	
	SR-5200	20	1.0			20-40k +0 −3		70	2.5/ 5.0		2.5	1.2			0.8	36	Mtr.	46	2	Yes	17 x 5 x 15	18	269.95	
	SR-800	40	0.5			20-25k +0 −3		60	2.5/ 5.0		1.8	1.1			0.8	40	Mtr.	43	2	Yes	18 x 5 x 13	26	329.95	
	SR-1100	55	0.5			20-35k +0 −3		70	1.8/ 4.0		1.6	0.8			0.8	42	Mtr.	65	2	Yes	18 x 5 x 13	26	419.95	
JVC	VR-5535	34	0.4	0.8	0.8	10-25k	10-50k ±1	65			2.0	2.0			0.4	35	2 Mtrs.	70	2	Yes	18⅝ x 12¾ x 6⅛	24.2	399.95	SEA tone contls.; hi & lo filters; mic mixing.
	VR-5525	23	0.8	0.8	0.8	10-25k	10-50k ±1	65			2.2	2.5			0.5	33	2 Mtrs.	60	2	Yes	18⅝ x 12¾ x 6⅛	22	349.95	SEA tone contls.; mic mixing.
	VR-5515	20	0.8	0.8	0.8	15-25k	10-50k ±1	65			2.2	2.5			0.5	33	2 Mtrs.	60	2	Yes	17¾ x 12¾ x 6⅛	21	279.95	Mic mixing; dual mtrs.
	4VR-5456 4-chan.	31	0.5	0.8		20-30k	10-50k ±1	65			1.8	1.5		0.4	0.6	38	2 Mtrs.	65	2	Yes	20 x 15⅞ x 7⅛	44	699.95	CD-4, matrix I, matrix II; fnt. & rear SEA; BTL power bridging.
	4VR-5446 4-chan.	22	0.5	0.8		20-30k	10-30k ±1	65			2.0	2.0		0.5	0.8	35	2 Mtrs.	65	2	Yes	20 x 15¾ x 6⅜	34.8	599.95	Same as above.
	4VR-5436 4-chan.	17	0.8	1.0		20-20k	20-30k ±1	65			2.0	2.0		0.5	0.8	35	2 Mtrs.	65	2	Yes	20 x 15¾ x 6⅜	31½	499.95	Same as above.
KENWOOD	KR 9340 4-chan.	45	0.5	0.5		10-60k	20-20k +1	60	1.5		1.8	1.0		0.3	0.5	40	Mtr.	80	6	Yes	21¼ x 14⅝ x 21¼		749.95	SQ, RM, CD-4; 2 4-chan. tape sys.; 2 4-ch. spkr. sys.; 4 VU mtrs.
	KR 8340 4-chan.	25*	0.8	0.8		10-60k	20-20k ±1	60	1.5		1.9	1.0		0.3	0.5	40	Mtr.	80	6	Yes	21¼ x 14⅜ x 6¼		619.95	*4-ch., 2-ch.: 70 W.; SQ, RM; opt. CD-4 adapt.
	KR 6340 4-chan.	17*	0.8	0.8		10-60k	20-20k ±1	60	1.5		2.0	1.5		0.4	0.5	40	Mtr.	70	1	Yes	19¾ x 5¾ x 14¼		489.95	Same as above. *4-chan.: 2-chan. 40 W.
	KR 5340 4-chan.	10*	0.8	0.8		10-60k	20-20k ±1	60	2.5		2.0	1.5		0.4	0.5	35	Mtr.	70	1	Yes	19¾ x 5¾ x 14¼		389.95	Same as above. *4-chan.: 2-ch. 25 W.
	KR 7200	55	0.5	0.5		10-30k	20-40k ±2	65	2.5		1.6	1.5	20-15k	0.4	0.6	40	Mtr.	75	2	Yes	17⅛ x 14 x 5¾	29	499.95	Dir. cplg.; DSD; dual protect crt.; dual tape sys.; triple tone contl.; mixing; A-B-C spkrs.
	KR 6200	45	0.5	0.5		13-30k	20-40k ±2	65	2.5		1.7	1.5	20-15k	0.5	0.6	40	Mtr.	65	2	Yes	17⅛ x 14 x 5¾	29	419.95	Dir. cplg.; dual protect crt.; triple tone contl.; dual tape; FET front-end, DSD.
	KR 5200	30	0.5	0.5		17-30k	20-40k ±2	65	2.5		1.8	2.0	20-15k	0.5	0.7	40	Mtr.	60	1	Yes	17 x 14 x 5	28	359.95	Dir. cplg.; DSD; dual protect. crt.; FET front-end; IC IF; linear dial; 2 sys. tape.
	KR 2300	14	1.0	1.0		30-50k		65	3		2.5	4.0		0.8	1.0	30	Mtr.	40	1	Yes	17¾ x 13⅞ x 5⅝	20.7	199.95	FET front-end; 3-stg. i.f.; terms. for phono, aux. tape; 2 spkr. sys.
KLH	54 4-chan.	25	0.5	0.5	0.3	16-30k	20-20k ±2	63	2.5	50	1.8	2	20-20k	0.3	0.8	35	2 Mtr.	45	2	Yes	17 x 14 x 5½	⅞	525.00	60W/chan. in 2-chan.; $10 more in west, south.
	55	12	1.0	0.5	0.5	20-20k	20-20k ±2	55	2.5	50	2.5	4	15-22½k	0.6	1.0	30	Mtr.	45	1	Yes	16 x 13 x 5¼	⅜	219.95	Price as above.
	52	33	1.0	0.5	0.3	15-30	15-22k ±2	69	3.5	60	1.8	2	15-30k	0.5	0.8	40	2 Mtr.	52	2	Yes	17 x 13 x 5¼	¾	319.75	Price as above.
	51	20	0.5	0.5	0.3	15-30k	10-35k ±2	63	2.5	140	2.5	2.5	10-35k	0.3	1.0	35	Mtr.	50	1	Yes	17 x 12 x 6¼	⅝	259.95	Price as above.
H. J. LEAK (ERCONA)	Delta 75	40					12.5-50k ±3 dB	-65	2.5		2.2	35	40-15k ±1 dB		0.25	35	Mtr.	40	2	Yes	16¼ x 12 x 5	25	595.00	

The Inside Story of Stereo Excellence



KENWOOD receivers, amplifiers, tuners, and 4-channel units are recognized the world over for their fine performance, value, and dependability. KENWOOD sound is crisp and clean, undistorted throughout the audio spectrum. KENWOOD design is amply luxurious with extra features at every price. The 'inside story' of such stereo excellence lies in KENWOOD's careful engineering that combines the most advanced technology with meticulous craftsmanship to give KENWOOD customers more stereo value, whatever their stereo budgets might be.

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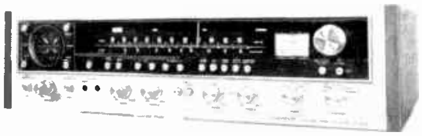
Receivers



Pioneer QX-949

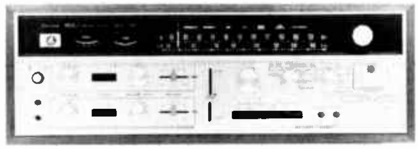
Notes: (1) All models solid-state except when model number is preceded by (T).

(2) "K" indicates kit price; "W" indicates wired price.

		AMPLIFIER									TUNER													
MANUFACTURER	MODEL	RMS Power/chan., W	THD at rated power, %	IM at rated power, %	IM at 1 watt, %	Power bandwidth, Hz–kHz	Frequency response, Hz ±2 dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	IHF sensitivity, µV	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %	THD stereo, 100% mod., %	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	SPECIAL FEATURES
LAFAYETTE	LR-4000 4-chan.	41	1.0			13-35k	20-20k ±1	70	4		1.65	1.5		0.15		40	Mtr.	60	2	Yes	21 x 13 x 5¼	40	569.95	Full-logic SQ, matrix, discrete; PLL MPX decode.
	LR-441 4-chan.	28	1.0			15-25k	20-20k	65	2.5; 6		1.65	1.5		0.1		40	Mtr.	60	2	Yes	19½ x 12⅔ x 5	35	479.95	As above.
	LR-221	18	1.0			30-35k	20-20k ±1½	62	2.5; 6		2.2	2.5		0.25		35	Mtr.	40	1	Yes	17¾ x 11½ x 4½	23	359.95	Full-logic SQ, matrix, discrete; Variblend.
	LR-2500 4-chan.	50	1.0			18-55k	20-20k ±¾	75	2.5; 6		1.65	1.5		0.1			Mtr.	60	2	Yes	19½ x 13¾ x 5¼	32	379.95	Derived 4-chan.; tone flat swit.; loudness; mute; hi fltr.; audio mute.
MAGNAVOX	KD1000	10	0.5	1.0	0.2	20-20k	20-30k ±1	60	2.5	60	4	4	20-15k	1.0	1.0	30	Mtr.	50	1	Yes	16 x 14 x 5	24	169.95	Also avail. w/top mntd. chgr.
	KD1001	18	0.5	1.0	0.2	20-20k	20-30k ±1	60	2.5	60	4	4	20-15k	1.0	1.0	30	Mtr.	50	1	Yes	16 x 14 x 5	26	199.95	Same as above.
	KD1002	25	0.5	1.0	0.2	15-20k	20-25k ±1	60	2.5	70	3	4	20-15k	1.0	1.0	30	Mtr.	60	1	Yes	18 x 16 x 5	35	229.95	Also avail. 32W./chan.
	KD-1004	50	0.5	1.0	0.2	15-20k	20-25k ±1	60	2.5	70	3	4	20-15k	1.0	1.0	30	Mtr.	60	1	Yes	18 x 16 x 5	40	319.95	Also avail. w/dig. freq. readout.
MARANTZ	19	50	0.15	0.15	0.15	8-80k	20-20k ±¼		1	100	1.7	1.5	30-15k ±½	0.15	0.15	45	Scope	80		No	18¼ x 16 x 6⅛	40	1200.00	'Scope display for tuning, multipath and audio.
	2270	70	0.3	0.3	0.3	7-70k	20-20k ±¼		1.8	100	1.9	1.6	30-15k ±1	0.15	0.25	42	Mtr.	80	2	Yes	16⅞ x 14 x 5	50	599.95	Multipath ind.; 2 tape recs.; pre-out, main in.
	2245	45	0.3	0.3	0.3	7-70k	20-20k ±½		1.8	100	2	1.6	30-15k ±1	0.15	0.3	42	Mtr.	60	2	Yes	16⅞ x 14 x 5	45	499.95	Pre-out, main in; front pnl. tape dub. jcks.; mdrng. tone contl.
	2230	30	0.5	0.5	0.5	15-50k	20-20k ±½		1.8	100	2	2.5	30-15k ±1	0.15	0.3	42	Mtr.	60	1	Yes	16⅞ x 14 x 5	40	399.95	Same as above.
	2220	20	0.9	0.9	0.9	15-50k	20-20k ±1		2.1	100	2.1	3	30-15k ±2	0.3	0.5	40	Mtr.	50	1	Yes	16⅞ x 14 x 5	34.6	299.95	2010 similar but 10W/ch., 20-20k bandwidth; $199.95.
	4300 4-chan.	40	0.15	0.15	0.15	7-70k	20-20k ±¼		2	100	1.9	1.5	30-15k ±1	0.2	0.3	40	Mtr.	70	2	Yes	18¼ x 16 x 6⅛	62.7	899.95	100W/ch. in 2-chan; Dolby; PLL FM demod.; Remote contl.
	4230 4-chan.	12	0.5	0.5	0.5	15-50k	20-20k ±1		1.8	100	2.8	2.5	30-15k ±2	0.6	1.0	32	Mtr.	48	2	Yes	16⅞ x 14 x 5	40	479.95	30 W/ch. in 2-chan., remote control.
ONKYO	TS-500	25 x 4	0.5	0.4		20-20k	20-30k	100	2.5	100	2	2	20-15k	0.4	0.8	40	Mtr.	65	1	Yes	21 x 14⅝ x 5½	35	N-A	Auto swit. for CD-4, SQ & RM.
	TX-440	28	0.5	0.4		20-20k	20-30k	65	2.5	100	2	2	20-15k	0.4	0.8	40	Mtr.	65	2	Yes	18½ x 14¾ x 5½	26	329.95	Dir. cpld. diff. amp crty.; s.s. & CT mtrs.; 2 tape mon. inputs.
PILOT	253	35	0.5	0.5		15-30k		75	2.5; 4.5		1.8	1.5	20-15k	0.4	0.8	36	Mtr.	65	1	Yes	18 x 13½ x 5⅛	30	299.90	FM tuning ind.; elect protect. circuit; large linear FM dial scale.
	254	65	0.4	0.5		10-40k		80	2.5; 4.5		1.8	1.5	20-15k	0.4	0.8	36	Mtr.	65	2	Yes	18½ x 17½ x 7	42	429.90	FM tuning ind.; large linear FM dial; access to main & preamps.
PIONEER	SX-828	60	0.5	0.5		10-60k			2.7		1.7	1.5		0.2	0.4	40	Mtr.	75	2	Yes	19 x 15 x 6	32	469.95	2 tape mon. w/dup., 2 phono inputs; FM & audio mut.; mic inputs; lo & hi fil.; FM linear scale.
	SX-727	40	0.5	0.5		10-60k			3.0		1.8	2.0		0.3	0.5	40	Mtr.	70	2	Yes	19 x 15 x 6	31	399.95	2 tape mon. w/dup; 2 phono in.; audio & FM mut.; lo & hi fil.; FM linear scale.
	SX-626	27	1.0	1.0		10-70k			2.5		2.0	2.5		0.4	0.5	40	Mtr.	70	1	Yes	18 x 14 x 6	22	339.95	2 tape mon. w/dup.; 2 phono in.; lo & hi fil.; FM muting.
	SX-525	17	1.0	1.0		15-30k			2.7		2.2	3.0		0.6	0.8	40	Mtr.	45	1	Yes	18 x 14 x 6	18	259.95	2 tape mon.; FM muting; FM noise filter; Loudness contl.

Receivers

Rotel RX-454



Sansui QRX-6500

Notes: (1) All models solid-state except when model number is preceded by (T).

(2) "K" indicates kit price; "W" indicates wired price.

		AMPLIFIER									TUNER												
MANUFACTURER	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 ± dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	IHF sensitivity, µV	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %	THD stereo, 100% mod., %	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	SPECIAL FEATURES
RADIO SHACK	QTA-790 4-chan.	36	1			18-45k	20-20k		2.5/5.0		2.0	1.5				35	Mtr.		1	Yes	18½ x 16 x 5½		598.00	SQ; wireless rem. FM tune & vol. contl.
	QTA-751 4-chan.	15	0.8			20-25k	20-20k ±1				3.3	2.5				30	Mtr.		2	Yes	13 x 19 x 5¾		299.95	SQ; synthesizer; wood case.
	STA-200	33	1			20-20k	20-30k	50	2.5/5.0		2.0	3.0				30	Mtr.		1	Yes	18½ x 13½ x 6¾		439.95	Wireless rem. on/off, FM tune, vol.; wood case.
	STA 150	32½	1			15-70k	20-20k		2.5/5.0		1.6					45	Mtr.		1	Yes	19¼ x 14⅜ x 5½		349.95	Auto tune; dub.; wood case.
ROTEL	RX150A	7.5	0.6	1.0	1.0	30-70k		60	2.8	70	5	7	30-20k		1.3	32	Mtr	25	1	Yes	16½ x 6⅝ x 5¼	10	149.95	2 spkr. sys.; tape mon.; tape dub. 1-2.
	RX800	40	0.2	0.2	0.1	5-50k		65	2.2	125	1.7	1.5	4-75k		0.5	38	Mtr.	70	2	Yes	16½ x 12 x 5½	26	439.95	Tape dub. 1-2; 2 phono; 2 aux.; hi-lo filter; aud. & FM muting; split power sply.; DCO circuit.
	RX154A 4-chan.	20	0.5	0.7	0.8	30-30k		60	3.0	70	4	7	20-50k		1.0	32	Mtr.	30	1	Yes	16¾ x 13⅜ x 5[illegible]	17	289.95	2-ch. 20W/ch. rms; SQ; matrix.
	RX 454A 4-chan.	45	0.1	0.2	0.1	20-50k		65	2.5	125	2.5	3	15-90k		0.5	40	Mtr.	45	1	Yes		26	479.95	45 W./ch. in 2-ch.; 45/45 stereo dual prog. source; SQ.
SANSUI	QRX 6500 4-chan.	37	0.5	0.5		20-30k	30-30k ±1.5		2.5		1.8	1.5		0.5	0.8	35	Mtr.	70	2	Yes	21⅛ x 14¼ x 7⅛	57	749.95	Vario-matrix decod./synth.; phase matrix.
	QRX 5500 4-chan.	30	0.3	0.3		10-30k	30-30k ±1½		2.5		1.9	2		0.3	0.5	40	Mtr.	60	2	Yes	23⅜ x 14½ x 8	47½	699.95	Same as above plus 4 VU mtrs.
	QRX 3500 4-chan.	22	0.5	0.5		10-33k	30-30k ±1.5		2.5	100	2.2	2		0.4	0.6	35	Mtr.	50	1	Yes	20¾ x 13⅞ x 7⅛	40.8	579.95	Vario-matrix decod/synth.; phase matrix; 3000 sim. but 15W/chan. & bandwidth 20-30k, $439.95.
	QR 1500 4-chan.	15	0.8	0.8		20-30k	30-30k ±2		3		3	3		1.0		35	Mtr.	45	1	Yes	19 x 12¼ x 5¼	19.8	339.95	500 sim. but 8W/chan. & bandwidth 30-30k, $229.95.
	8 Deluxe	60	0.2	0.3		5-40k	5-50k +0.5 −1		2.5		1.7	1.5	30-12k	0.3	0.5	35	Mtr.	80	2	Yes	17⅝ x 13 x 5½	42	599.95	4-chan. ready.
	771	40	0.5	0.5		15-40k	20-40k ±2		2.5		2	2	30-12k	0.4	0.6	35	Mtr.	60	1	Yes	18⅞ x 12¾ x 5⅜	26½	339.95	4-chan. ready; 661 sim but 25W/chan., $289.95.
	350A	20	1	1		30-30k	30-30k ±1		2.2		3	3		1.0		30	Mtr.	32	1	Yes	16¼ x 12 x 5⅝	21	239.95	
SANYO	DCX 2700K 4-chan.	30	1.0	1.0	0.5	30-30k	30-30k +0 −3	60	5	150	2.7	2.5	25-30k	1.0	1.0	30	Mtr.		1	Yes	18 x 14¼ x 5	15	269.95	Matrix & SQ; 2/4 chan. tape inputs/outputs.
	DCX 3000KA 4-chan.	40	0.5	1.0	0.5	25-40k	25-40k +0 −3	60	5	150	2.2	2	25-40k	0.8	0.8	30	Mtr.		1	Yes	18⅝ x 11⅞ x 4¾	16½	299.95	2 Matrix, SQ, discrete; indiv. & master vol. contls.
	DCX 3100K 4-chan.	60	0.5	1.0	0.5	25-45k	25-45k +0 −3	60	3	150	2.2	2	20-45k	0.8	0.8	30	Mtr.		1	Yes	19½ x 13¾ x 5	17½	349.95	Same as above plus amps may be "strapped" for 2-ch. stereo.
	DCX 3300KA 4-chan.	80	0.5	1.0	0.5	20-50k	20-50k +0 −3	60	3	150	2	2	18-50k	0.8	0.8	30	Mtr.		1	Yes	19⅜ x 12¾ x 6	30	449.95	2 Matrix, SQ, discrete; indiv. & master vol. contls.; 4 output pwr. mtrs.
	DCX 3500K 4-chan.	80	0.5	1.0	0.5	20-50k	20-50k +0 −3	60	3	150	2	1.5	18-50k	0.5	0.8	33	Mtr.		1	Yes	19⅜ x 12¾ x 6	31	549.95	4 output pwr. mtrs.; matrix, SQ & CO-4; amps have strapping cap.

The 'Best Buy' Line

S-7050 "**The winner in our evaluation**," Music World. (36 watts IHF; 10 + 10 RMS [8 Ohms @ 1 KHz]; 3.5 μv FM Sensitivity [IHF]; 40 dB selectivity)

S7100A. "**Best Buy**," a leading consumer testing magazine. (22 + 22 watts RMS [8 Ohms @ 1 KHz]; 1.9 μv Sensitivity [IHF]; 50 dB selectivity)

S7200. "**Best Buy**," a leading consumer testing magazine. (40 + 40 watts RMS [8 Ohms @ 1 KHz]; 1.8 μv FM Sensitivity [IHF]; 60 dB selectivity)

S7900A (AM/FM) & S8900A (FM only). "**Best Buy**," a leading consumer testing publication. (60 + 60 RMS [8 Ohms] 20 – 20,000 Hz; 1.7 μv sensitivity [IHF]; 65 dB selectivity)

There are certain rewards for producing the best receivers in this business.

One of them is critical acclaim.

And we admit, that when a leading consumers' testing magazine picks three of our receivers as "Best Buys" and another independent publication rates our S-7050 as the best of the low-priced receivers, to us it's like getting an Oscar.

But nothing is more exciting than being discovered by thousands of new consumers who choose Sherwood over the giants of the industry.

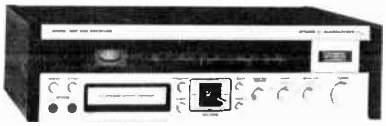
And, this year, as never before, the word is getting around.

Sherwood Electronic Laboratories
4300 N. California
Chicago, Illinois 60618

Sherwood

The word is getting around.

Receivers



Superscope QRT-440

Technics by Panasonic SA-8000X

MANUFACTURER	MODEL	AMPLIFIER: 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 ? dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	TUNER: IHF sensitivity, µV	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %	THD stereo, 100% mod., %	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	SPECIAL FEATURES
SHERWOOD	S-7050	10	0.9	1.0	0.35	15-50k	30-20k ±1	65	2	50	3.5	4.0	20-15k ±2	0.6	1.0	35	Mtr.	40	1	Yes	16 x 12 x 5½	24	174.95	Wood case; A, B spkr. swit.
	S-7100A	20	0.9	1.0	0.35	15-50k	20-20k ±1	65	1.5	60	1.9	2.8	20-15k ±1	0.5	0.8	40	Mtr.	50	1	Yes	17½ x 13½ x 5⅝	30	219.95	Wal. case; tape dubbing.
	7200	40	0.7	0.7	0.25	12-35k	20-20k ±0.5	60	2.2	60	1.8	1.9	20-15k ±1	0.25	0.5	40	Mtr.	60	1	Yes	17½ x 14 x 5¾	32	339.95	Wal. case; Tape dubbing; 4-chan. switching.
	SEL-200	60	0.2	0.6	0.1	8-35k	20-20k ±0.5	65	1.7; 3.6; 8.2	100	1.5	1.7	20-15k ±1	0.15	0.25	40	2 Mtrs.	70	2	No	18¾ x 13 x 5¾	37	599.00	LeGendre torroidal FM i.f. filters; FET mash ckt.
SONY	STR-6036A	18	0.8	0.8	0.1	10-25k	30-40k ±2	60	2.5	60	2.2	1.5	30-15k ±1	0.3	0.8	35	mtr	60	1	yes	17⅛ x 13½ x 5¾	19	229.50	15 + 15 W. at 20-20k; wood. cab.; dir. spkr. cplng.
	STR-6046A	22	0.8	0.8	0.1	10-25k	20-40k ±2	60	2.5	60	2.2	1.5	30-15k ±1	0.3	0.8	35	mtr	60	1	yes	17⅛ x 13½ x 5¾	19¾	279.50	20 + 20 W. at 20-20k; wood cab.; dir. spkr. coupling.
	STR-7045	40	0.2	0.2	0.1	15-30k	10-60k ±2	70	1.8	70	2.6	1.5	20-15k ±1	0.2	0.5	38	mtr	70	1	yes	18¼ x 14¾ x 6¼	31	349.50	30 + 30 W. at 20-20k; wood cab.; dir. spkr. coupling.
	STR-7055	45	0.2	0.2	0.1	15-35k	10-70k ±2	70	2.0	70	2	1	20-15k ±1	0.2	0.5	38	mtr	70	1	yes	18¼ x 14¾ x 6¼	33⅓	429.50	35 + 35 W. at 20-20k; wood cab.; dir. spkr. coupling; 2 tape mon.
	STR-7065	70	0.2	0.2	0.1	15-35k	10-70k ±2	72	2.0	70	2	1	20-15k ±1	0.2	0.5	38	mtr	70	2	yes	18¼ x 14¾ x 6¼	33¼	529.50	60 + 60 W. at 20-20k; wood cab.; dir. spkr. coupl.; 2 tape mon.
	STC-7000		0.1	0.1			10-100k ±1	72	2	70	1.7	1	30-15k ±1	0.3	0.5	40	mtr	100	2	yes	18¾ x 13¾ x 5⅞	22½	589.50	Pwr. amp not incl.; 2 tape mon.
	SQR 6650 4-chan.	25	0.8	1.0	0.5	10-40k	20-50k ±2	60	2.5	60	2.2	1.5	30-15k ±1	0.3	0.8	35	mtr	70	1	yes	17⅛ x 13½ x 5¾	21½	329.50	8+8+8+8 in 4-ch.; built-in SQ; 4 VU mtrs.; dir. spkr. coupling.
SOUNDCRAFTSMEN	2000A 4-chan.	20	0.2	0.4	0.08	25-30k	15-50k ±1	65	3		2.2	3	20-15k	0.6	0.9	36	Mtr.	45	1	Yes	17⅛ x 12½ x 5	16	299.50	Matrix; wal. case incl.; 120-240 V.
	3000A 4-chan.	30	0.2	0.2	0.07	15-45k	9-100k	67	2.5		2.0	2	15-15k	0.3	0.5	38	Mtrs.	50	2	Yes	17⅛ x 12½ x 5	21	349.50	Same as above, plus splt. pwr. sply; 3 phono inputs; 2 tape inputs; dir. cpld.
SUPERSCOPE	R-330	9	1.0			30-30k			2.5		5	5	30-15k ±3	1.0	1.0	32	Mtr.		1	Yes	23 x 19 x 12¾	19	179.95	Tape mon.; loudness; 4-ch. FM output; flywheel tuning.
	R-350 4-chan	25	1.0			15-40k			2.5		2.8	3	20-15k ±1.5	0.6	1.0	32	Mtr.		1	Yes	23 x 19 x 12⅝	25	279.95	Flywheel tun.; FM mut.; loudness; mono; tape mon.; Quadraphase; slide bal.; 4 ch. FM output w/Hi filt. swit.
	QRT440 4-chan.	12*	1.0	1.0		20-20k			2.2		5	5	20-12k ±1.5	1.0	1.5	32	Mtr.		1	Yes			349.95	*2-ch., 6W./chn in 4-ch. mode; 2/4-ch. 8-tk play.; SQ amb., discrete; joy stick bal. contl. (4-ch); flywheel tuning.
TANDBERG	1020A	40	0.2	0.2	0.1	7-30k	12-70k	76	2-8	100	2	0.9	20-15k	0.2	0.3	40	Mtr.	80	2	Yes	17 x 12 x 5¼	21	499.90	Pwr. ind.; input lev. contl. for phone, tape 1 & 2; tape 3 preamp rec.; elec. tune; 2 pwr. supply.
	1055	55	0.2	0.2	0.1	6-40k	12-70k	76	2-8	100	2	0.9	20-15k	0.2	0.3	40	Mtr.	80	2	Yes	17 x 12 x 4¼	23	579.90	Same as above.
TELEDYNE (OLSON)	RA777	25	0.2	0.75	0.6	10-40k	8-50k	65	2.2	12.0	3.5	2	20-20k	0.5	0.9	35	mtr	65	2	yes	15¼ x 12¾ x 4½	18	269.98	
	RA632 4-chan.	15	0.5	0.85	0.7	15-26k	19-23k	62	2.5	25	2	3	17-20k ±1	0.6	1.0	35	mtr	62	1	yes	18 x 10 x 3½	16	259.98	
	RA618	15	0.5	0.85	0.7	15-26k	19-23k	62	2.5	25	2	3	17-20k	0.6	1.0	35	mtr	62	1	yes	18 x 10 x 3½	16	189.98	
	RA660 4-chan.	5	0.5	0.9	0.72	20-24k	25-30k	60	2.8	25	3.0	3.5	20-20k	0.75	1.2	30	mtr	62	1	yes	16 x 10 x 3½	14	214.98	

A tuner and an amplifier should be mated somewhat more carefully than a husband and wife.

Marry a tuner with an amplifier, and you have not only a receiver, but a union that's truly indissoluble.

It had better be a good one.

For to join a fantastic tuner with a lesser amplifier, or a sensational amplifier with a tuner that's merely great, is to invite unhappiness. But match two equals, and they can make beautiful music together—as they must, to live in happy harmony with you.

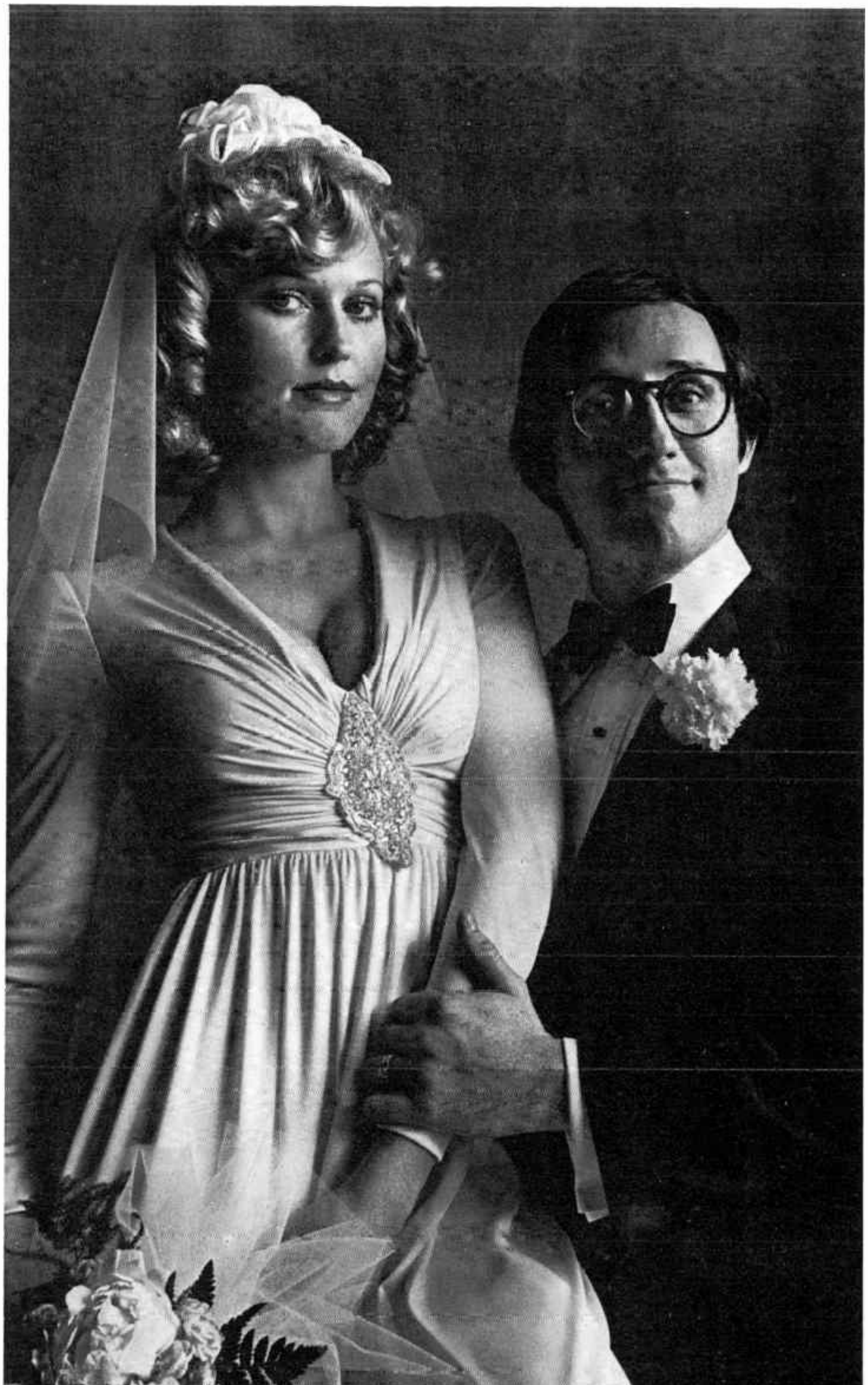
That's why the Sony STR-7065 receiver is a perfect mating. Its tuner has the sensitivity to reach out for signals from even the most distant fringe locations, yet has discrimination enough (70dB IHF selectivity, 1dB capture ratio) to pluck one signal clearly from a crowded band.

And the 7065's direct-coupled amplifier brings to this union the strength of 60+60 RMS watts (from 20Hz to 20kHz at 8 ohms with less than 0.2% distortion). Versatile controls offer a choice of three speaker pairs, mic mixing and dual tape monitors. Switchable preamp-out/amplifier-in connections permit independent use of each section and addition of accessories.

Such a happy union should sparkle visibly as well as musically. And the 7065 does, with lights that tell which of its many functions are in use and dual tuning meters.

The Sony 7065, at $529.50,* is our top-of-the-line receiver. For those who feel a union can survive with fewer luxuries (no indicator lights, signal strength meter or mic mixing control), we offer the Sony STR-7055. It has 35+35W RMS, 20Hz to 20kHz at 8 ohms with 0.2% THD. At $429.50* it's an equally well-mated receiver. Sony Corporation of America, 9 West 57th Street, New York, N.Y. 10019.

SONY®



*Suggested retail. Prices include walnut finish cabinets.

Receivers

MANUFACTURER	MODEL	AMPLIFIER: 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 ±dB	Rated output S/N, Phono, dB	Phono sensitivity, mV	Phono overload, mV	TUNER: IHF sensitivity, µV	Capture ratio, dB	Frequency resp., Hz	THD Mono, 100% mod., %	THD stereo, 100% mod., %	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	SPECIAL FEATURES
TECHNICS BY PANASONIC	SA-6800X 4-chan.	25	0.5	0.7		7-40k	10-70k +0 −1	73	3		1.8	1.5		0.4		35	2 Mtrs	70	2	Yes	16 x 17½ x 6¼	35	599.95	AFD matrix decoder w/sel. phase shift, matches any sys.; rem. bal. contl.; mic input; 3 tape mon.
	SA-6700X 4-chan.	23	0.5	0.7		7-40k	15-50k +0 −3	70	2.5		1.8	1.5	20-15k	0.3	0.4	40	Mtr. & Scope	65	1	Yes	16½ x 17½ x 6¼	35	569.95	AFD matrix decoder (as above); BTL pwr. amp brdg.; 4 VU mtrs.
	SA-8000X 4-chan.	16	0.5	0.7		5-40k	10-50k +0 −3	70	1.5		1.9	1.8	20-15k	0.3	0.4	40	Mtr.	65	5	Yes	19½ x 15¾ x 6¼	35	499.95	CD-4; AFD matrix decoder; BTL pwr. amp brdg.; 4 VU mtrs.
	SA-5400X 4-chan.	8	0.8	1.0		7 28k	10-50k +0 −3	70	2.5		2	2	20-15k	0 4	0.7	37	Mtr.	65	1	Yes	16¼ x 13¼ x 5½	20	269.95	2 matrix decoders; BTL pwr. amp brdg.; tape mon.; opt. rem. bal contl.
	SA-5200	13	0.8	1.0		7-30k	10-50k +1 −3	70	3		1.9	2	20-15k	0.4		35	Mtr.	60	1	Yes	16⅛ x 14¼ x 5¼	18	199.95	Matrix; 2 tape mon.; dir. cpld. output.
V-M	1545 4-chan.	4	5.0				70-10k	55			15		30-15k			20	Mtr.		1	Yes				E-V decoding. 1548 same W. 8-Tk player.
YAMAHA	CR 400	16	0.5	0.5	0.1	15-50k	20-50k +0.5 −3	72*	3	135	2.5	2.0	20-15k ±3	0.3	0.8	40	Mtr.	65	2	Yes	17 x 6 x 12	18	299.95	*IHF A ntwrk.; LED ind.; mic mix.; OCL built-in filter.
	CR 600	30	0.1	0.1	0.05	5-70k	10-50k +0.5 −1	75*	3	135	2.0	1.5	20-15k ±2	0.3	0.5	40	Mtr.	75	2	Yes	19 x 6 x 12	29	449.95	*Same as above; auto-tch. AFC; cont. loud. contl.; 2 tape input.

THE EVOLUTION OF FOUR-CHANNEL EQUIPMENT

Leonard Feldman

The 4-channel era is less than four years old and we have already witnessed at least four generations of equipment suitable for home reproduction of surround sound. There seems to be a frenetic endeavor on the part of component high fidelity manufacturers to be "first" with innovative products, and it is this very desire to reach the marketplace ahead of all competition that has given rise to a sometimes confusing array of add-ons, conversion components, interim components and finally, "universal" products which purport to solve all four-channel needs —now and in the future.

The transition from monophonic to stereophonic sound which, from the audiophile's point of view, took place some ten or twelve years ago had fewer "false starts," viewed in retrospect. Few stereo converts ever bothered to add a second amplifier component equipped with any sort of "combining" controls. Typically, the consumer who wanted stereo sound simply went out and bought a two-channel receiver or a two-channel integrated amplifier as the "first phase" of his transition. Later, when stereo FM broadcasting began, he either added a mutliplex adapter (these devices were popular for only two or three years) or, if he was fortunate enough to have separate amplifier and tuner facilities, he disposed of his single-channel tuner and purchased a stereo tuner. The age of solid-state equipment arrived at a very opportune moment in terms of home stereo systems, since anyone who owned a mono tube-type receiver was provided with additional incentive to buy an all-in-one solid-state stereophonic receiver. He thereby acquired state-of-the-art electronics and stereo reproduction with one new investment.

From the time stereo conversion was completed, a measure of stability descended upon the industry. Equipment up-dating from the mid-sixties to the present was largely confined to improved performance rather than change of basic formats.

Four-Channel—Phase One

The first, hesitant entry into four-channel equipment on the part of the buying public is best characterized by the block diagram of Fig. 1—a diagram which appeared in literally dozens of tutorial articles and on countless manufacturers' brochures. Users were advised to buy a second stereo amplifier (its power output capability was the subject of much debate), a second pair of speakers (their similarity to the first pair is *still* a subject of debate) and some form of matrix decoder with which to tie the whole thing together. At just about the same time, two additional alternatives were offered. There were the early quadraphonic amplifiers, which were simply four amplifier channels mounted on a single chassis with *no* matrix decoding facilities. There were also a group of decoder/amplifier products which offered one or more matrix decoder circuits *plus* a pair of amplifying channels and a master volume control which controlled the level of all four channels simultaneously. Some of the early decoder add-ons recognized the importance of this control and included it in their products as well. The four-channel amplifiers sans decoding facilities have just about vanished from the scene, since they are neither fish nor foul. Owners of stereo systems would find no need for them, and newcomers desiring four-channel sound at the outset certainly had no desire to have to add a separate decoder after making a heavy investment in a four-channel integrated amplifier. Decoders and decoder/amplifiers still abound, but their character and complexity have been altered considerably since those first, simple matrix "black boxes" were first offered to the public.

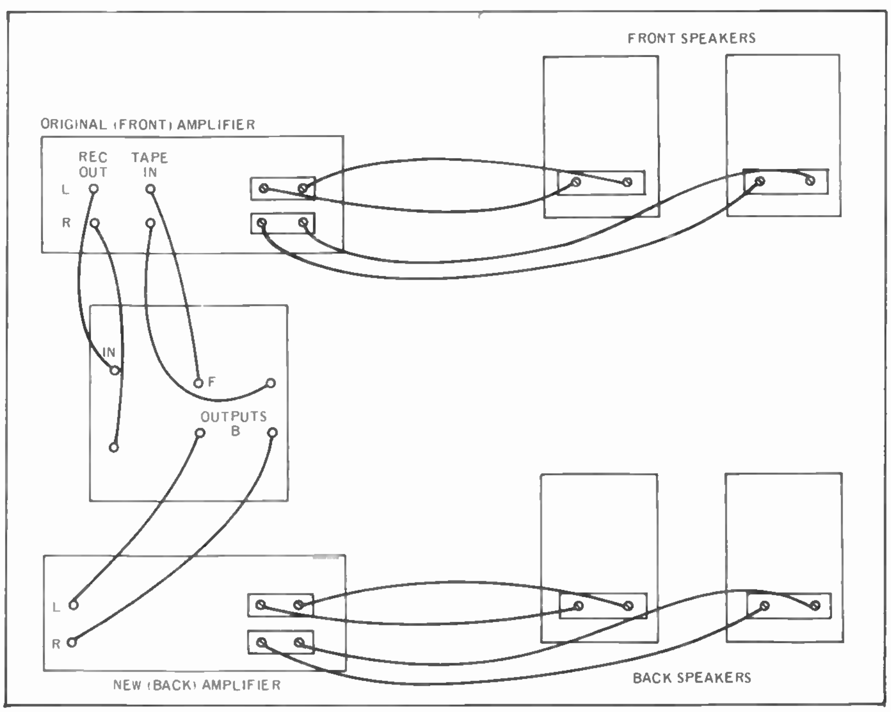


Fig. 1—"Classical connection diagram for connecting matrix decoder via tape monitor jacks of existing amplifier or receiver.

Four-Channel—Phase Two

Early auditioners of four-channel sound were subjected to two kinds of public demonstrations. There was, of course, discrete four-channel programming on open-reel tape. RCA adopted a position that any four-channel programming offered by them would have to be discrete (as opposed to "matrix"), with "full separation." Since the viable discrete disc was still a development of the future, RCA introduced Q-8 cartridges, similar in form to the popular 8-track cartridges which had gained popularity in automobile and home use. Despite their limited signal-to-noise and frequency response, the Q-8 format taught the four-channel listener that "discrete" channel separation was audibly superior to the rather minimal separation achieved by most simple matrix systems.

There then began a race on the part of many manufacturers to introduce "second generation" matrix decoders which included "logic circuitry." Logic, or gain-riding circuitry, simply senses which channel is instantaneously dominant and either increases the gain in that channel or reduces the gain in non-dominant channels (or performs some of each gain change at once). The result is improved apparent separation for most musical situations. The most popular matrix method is that proposed by CBS and called SQ. Another matrix system vying for consideration is QS—proposed by Sansui Corporation of

Japan. This latter form of matrix conforms to the so-called "regular matrix" sound field adopted by many other Japanese manufacturers. What the two systems have in common, however, is that they offer an inherent separation limitation of only 3 dB of actual separation. In the case of CBS-SQ, left to right separation is fully maintained, while front-to-back and diagonal separation is limited to 3 dB in their simple matrix system. In the case of Sansui QS matrix discs, diagonal separation is total, while adjacent flanking channels are limited to only 3 dB of separation. CBS's dominance in the software field resulted in the availability of hundreds of SQ discs which, if played on simple SQ decoders, left much to be desired in the way of separation. QS decoders were supplied by Sansui to many radio stations and recording studios as well, so that the preponderance of four-channel program material suffered from "lack of separation" if played on simple decoding equipment.

Sansui offered its "variomatrix," a sophisticated "logic circuit" arrangement designed to offset these limitations in separation, while CBS, through its hardware manufacturing licensees offered first "front-back" enhancement circuits and, finally, "double logic" circuits which accentuated both front-back separation and side-to-side separation when required. Thus, a whole new breed of "matrix decoder" products appeared upon the market and those listeners who had hastily purchased "simple" matrix decoders found that their very recent purchases no longer represented the latest four-channel technology.

During this same "phase two" period, receiver manufacturers quickly designed and developed integrated four-channel receivers which included one or more simple matrix decoders. Most featured at least three-position switches with settings for SQ, Regular Matrix and "Stereo Enhancement" (a matrix similar to the original Electro-Voice proposal which offered greatest front-back separation and was therefore particularly effective in keeping "front-and-center" vocalists up front, where they belong, while permitting out-of-phase random signals of stereo discs to wander around towards the rear to simulate a four-channel effect). Most of these receivers offered moderate power output capabilities—around 10 to 20 watts per channel was typical. Despite the recent flurry of high-powered stereophonic receivers on the market, the lower powered quadraphonic units were justified by their manufacturers because, after all, with four channels operating simultaneously there was that much more acoustic power being pumped into the listening room. The obvious flaw in this argument occurs if a listener prefers to play some of his program material in two-channel stereo, in which case two of the four amplifying channels simply idled along and did nothing. The wastefulness of this arrangement must have become obvious to manufacturer and user alike very early in the four-channel evolution and undoubtedly accounted for new system purchasers' reluctance to invest in these new receivers. Many listeners felt that the amount of program material then available for four-channel listening was at best limited and wanted to start their systems by purchasing only two speakers. These buyers were in somewhat of a bind, in that they were reluctant to purchase stereo receivers as well, recognizing that it would not be long before they would, indeed want to "switch" to four-channel." The idea of then having to add a decoder and a second amplifier did not appeal to such new purchasers. They felt that if they were just starting to assemble a system that they should certainly not have to "add on" and "make do" in just a few months. As a result, the "brute force" and obvious type of four-channel receiver did not enjoy the success its manufacturers had hoped for and the anticipated race towards four-channel slowed down to a veritable crawl!

Four-Channel—Phase Three

It would be difficult to assign the credit for the first two/four channel "bootstrapped amplifier" receiver to a single high fidelity component manufacturer. So as not to become involved in the argument as to who was first, let's assign the credit to Bell Laboratories who some years ago published a technical paper describing a method of connecting two solid state amplifiers in a bridge-like configuration to obtain more than twice the power output capabilities of each. Whether it was by grapevine communication or industry-wide inspiration, several manufacturers, almost simultaneously, designed and produced a new kind of quadraphonic receiver which offered distinct advantages to the perplexed audiophile.

For the hesitant quadraphonic equipment buyer, the new receivers offer full-powered stereo, with four amplifier channels bridged or combined to provide higher-powered two-channel operation. Thus, the purchaser who wants to begin his home system by purchasing two speakers is secure in the knowledge that half his power output capability is not being wasted. Then, when he's convinced that four-channel is here and that there's enough happening by way of program material, broadcasts and the like, he can purchase that second pair of speakers, flip a switch and, like biological cells, the two amplifiers divide into four, albeit at somewhat reduced total power. Typically, such a receiver producing about 50 watts per channel in the stereo mode would be expected to deliver about 20 watts per channel when the quadraphonic switch is thrown. Naturally, all of these receivers contain matrix and control facilities similar to their less flexible predecessors, and that brings us to what we hope is the *final* phase in this quadraphonic equipment revolution.

Four Channel—Phase Four

While these hectic three phases of equipment development underwent their gestation and production periods, the people who gave immortality to a little dog listening to an acoustic phonograph horn (and have since deserted "little nipper" in favor of a more avant garde corporate image, much to the distress of nostalgia buffs such as myself) have not been idle. Having put their money on the "discrete" four-channel approach, they huffed and they puffed and finally declared that the discrete disc was "ready." It turned out to be none other than the CD-4 disc which had been developed by Japan Victor Company of Japan and which had been briskly selling in the Orient for nearly two years. RCA, however, improved, refined, perfected and renamed the disc—and now we have quadradiscs, plus the need for a new kind of decoder called a demodulator. It appeared for a while that "Phase Four" would consist of the addition of yet another "black box" and the need for six more audio pin-to-pin cables, in order to hook-up for Quadradisc playback, as shown in the block diagram of Fig. 2. Furthermore, RCA grudgingly admitted that in *most* instances a new phono cartridge and stylus would be required if the high frequency content of these new Quadradiscs was to be properly traced. No mention was made of the fact that the new "demodulator" included low-level preamplification circuitry, thereby obsoleting the preamplifier section of one's existing receiver or amplifier, but this is apparent from the connection arrangement shown in Fig. 2. In short, if you were a four-channel pioneer dating back to "phase one," you might have ended up with a total system

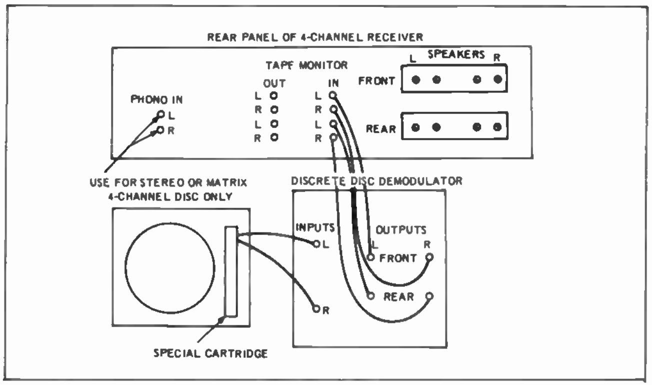


Fig. 2—Most "4-channel receivers" offered to date still require an externally connected Quadradisc demodulator if RCA discs are to be played

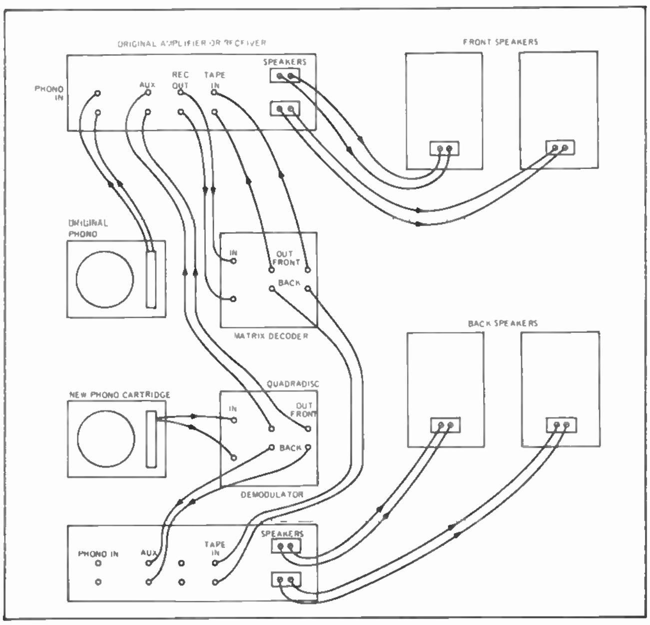


Fig. 3—Anyone keeping pace with 4-channel equipment from "the beginning" would end up with the components shown above.

shown in the block diagram of Fig. 3—heaven help you!

Well, for the moment, the only separate demodulators offered for sale are those made by JVC and Panasonic (who endorses the "discrete" approach along with RCA), and in view of what seems to be happening in "phase four," such separate demodulators are likely to be around for an even shorter period than the "matrix decoders" of "phase one" and "phase two." The new crop of receivers, as you may have guessed, includes (at very least) a *four* position switch for Quadradiscs, SQ-discs, Regular Matrix and, in some cases, 2-channel enhancement. With this arrangement, there is a minimum of circuit redundancy and the electronics of your system settles down to one all-inclusive piece (or, at most two if you prefer a separate tuner) once again.

Future Phases of Four Channel

Before you breathe a sigh of relief, you had better know that it's not all over just yet. For one thing, there remains the question of "logic circuitry." Some of the new receivers equipped with both matrix and "discrete" demodulator circuits will still lack sophisticated logic circuitry for use in their matrix modes. The use of such receivers is likely to give RCA a much needed boost, because when discrete discs are compared with matrix discs played through simple matrix decoders there's no question about the superiority of the quadradisc (if we confine the comparison to separation or image placement). Some receiver manufacturers, therefore, are going all the way and including both matrix-logic circuitry and quadradisc demodulator circuits in their latest products and such products (however costly they must be) are sure to become "favorites" in the immediate future. In time, the "strapping" feature will no doubt be dropped, as buyers are increasingly convinced that the only way to listen to music is in four-channel surround sound. Elimination of the "bootstrapping" feature may partly offset the cost of including logic-matrix *plus* quadradisc playback capability.

Lurking behind the scenes, however, is one more technological breakthrough—and that has to do with four-channel FM broadcasting. Obviously, the many stations currently featuring quadraphonic programming are confined to one or another matrix system. Since only two channels (however encoded) need be transmitted for this format, present FCC broadcast rules relating to stereo broadcasting are adequate and need not be modified. If past performance is any guide, the purists among us will not settle for this form of four-channel broadcasting forever and sooner or later there will be an approved method of discrete four-channel FM transmission. Committees are already at work on the problem and some seven proposed systems are currently under consideration. It's likely to take at least several more years before the FCC gives the nod to one of these systems (or to an as yet unproposed system), but when they do, you'll no doubt have to run right out again for still another black box—for which a back-panel jack is already being provided on today's receivers and tuners. You can be sure that the progressive and eager manufacturers in the high fidelity component industry would add the needed circuitry for four-channel discrete reception *right now*—if only they knew which circuits to add. Since that depends upon a government ruling, however, you'll have to be content with the equipment you purchased during the first four phases of the quadraphonic equipment revolution.

Actually, the stages in this evolution are nothing for the dedicated audiophile to grumble over. What sort of a hobby would this be if new and exciting equipment failed to come upon the scene every couple of years? At least model changes in *this* industry result in better sound—rather than in just fancier front panels. . . . Æ

Manual Turntables & Tonearms

AR XB

Pioneer PL-61

Empire Troubador 598II

Dual 701

SPEEDS (see letter code)

A—33, 45, 78
B—33, 45
C—33 only
D—16, 33, 45, 78
E—16, 33, 45
F—cont. variable

		TURNTABLES										TONE ARMS												
MANUFACTURER	MODEL	Speeds (see letter code)	Wow & Flutter at 33⅓, %	Rumble (NAB) dB	Motor type	Platter diameter, in.	Platter weight, lbs.	Drive	Arm mounting provision	Dimensions, W x D x H, in.	Weight, lbs.	MODEL (for separate arms)	Overall length, in.	Pivot-stylus dist., in.	Vertical bearing	Lateral bearing	Stylus force method	Max. tracking error, deg.	Cartridge weight range, gms.	Arm resonance, Hz	Stylus force range, gms.	Weight, if sep., oz.	Price	SPECIAL FEATURES
ACOUSTIC RESEARCH	XB	B	0.5	*-38	Syn.	11¾	4	Belt	Integ.	16½ x 12¾ x 5½	13½		12	9	Cone point	Ball	Bal.	0.35		10-15	0.5 8.0		109.95	*CBS RRLL, -58; w. stylus frc. gauge, base, cover; damped cue. XB-91, w. Shure M91-ED cart., $164.90. XBU, 110-120V; 50-60 Hz, $106.95.
AUDIO RESEARCH												TA-1	11¼	9	Ball	Ball	Bal.	2.0	3-15	8	0-6		175.00	Ultra-low spur. resp.
DECCA (PAOLI)												Int'l	11¼	9	Unipivot	Unipivot	Bal.		2-20	14	0-4		130.00	Visc. damp.; Adj. mag. anti-skate. Decca lift, $35.00.
DUAL (UNITED AUDIO)	701	B	0.03	-70	Elect.	12	9.7	Direct	Integ.	16.5 x 14.4 x 5.8	24		12¾	8¾	Ball	Ball	Bal. & spg.	0.16	2-10	*	0-3		350.00	*2 reson.-cancel. filters; twin gimbal tonearm susp.; self-reg. elect. speed mon.; pitch contl.; ill. strobe; auto start & stop.
EMPIRE	598 II	B	0.01	-70	Hys.	12	7	Belt	Integ.	17¼ x 15¼ x 8	30	990	12	9	Ball	Ball	Calib. Bal. & spg.	0.7	4-14	6	0-6		349.95	Incl. 1000ZE/X low trkg. cart.
HITACHI	PS-33	B	0.1		Hys.	12	2½	Belt	Static Bal.	19 x 7 x 15					Ball	Ball	Wt.				0-4		249.95	
JVC	VL-8	B	0.05	-60	Servo	12	4	Belt	Integ.	19¾ x 16¼ x 7½	22		13	10	Ball	Ball	Bal.	0.94	5-20	7	0-4		259.95	CD-4 ready.
	VL-5	B	0.1	50	Syn.	12	2	Belt	Integ.	18½ x 14½ x 7⅜	15		12	9	Pivot	Ball	Static bal.		5-25	10	0-3		109.95	CD-4 ready; arm lifts at end of record.
LENCO (BENJAMIN)	L75	F	0.07	-38	Ind.	11¾	8.8	Idler	Integ.	17½ x 13¾ x 6¼	32			8	Knife edge	Ball	Bal.				0-5		109.50	Anti-skate adj.; Infinite speed adj.
	L78	F	0.06	-40	Ind.	11¾	8.8	Idler	Integ.	17¼ x 13¾ x 6¼	32			8	Knife edge	Ball	Bal.				0-5		179.50	Auto shutoff & arm lift; anti-skate.
	L85	B	0.06	-45	Syn.	12½	3.6	Belt	Integ.	18½ x 14⅜ x 3	30			8¾	Ball	Ball	Bal.	0.6			0-5		249.50	± 3% speed var.; auto shutoff & arm lift; Ill. strobe; anti-skate adj.
PHILIPS	GA-212	B	0.1	-40	d.c.	11½	6	Belt	Integ.	15¼ x 13¾ x 6¼	13¼		10½	8¾	Pin & Bush.	Ball	Bal.	0.7	0-18	7	¾-4		174.50	
	GA-407	B	0.2	-35	syn.	11½	6	Belt	Integ.	12¼ x 10½ x 6	12						Bal.		6-18	7	1-4		119.50	
PIONEER	PL-61	B	0.05	-55	DC servo	12¼	4	Belt	Integ.	19 x 17 x 7	24												299.95	Built-in strobe; anti-skate; var. speed contl.; dust cover.
	PL-51	B	0.06	55	DC servo	12¼	3	Direct	Integ.	18 x 16 x 7	23												249.95	DC direct drive; anti-skate; var. elec. speed contl., dust cover.
	PLA35	B	0.1	-47	Syn.	12	3	Belt	Integ.	19 x 16 x 7	20												149.95	2 mtrs.; fully auto. or manual; dust cover.
	PL-12D	B	0.10	-47	Syn.	12	3	Belt	Integ.	17 x 14 x 6	16												99.95	Built-in overhang checker; var. anti-skate; dust cover.
RABCO	ST-4	B	0.08	-60	Sync.	11		Belt	Integ.	15¾ x 18 x 6	15	SL-8E	14		Cone	Cone	Bal.	0			¼ min.	36	199.00	Straight line trkg.; elec. servo.

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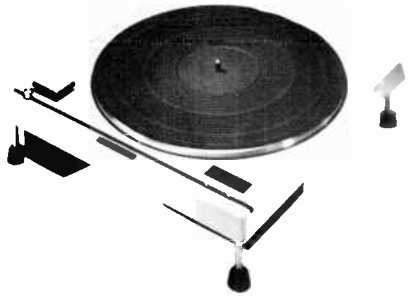
THORENS

Manual Turntables & Tonearms

Thorens TD-125AB Mk II

Yamaha YP-700



Technics by Panasonic SP-10

SPEEDS (see letter code)

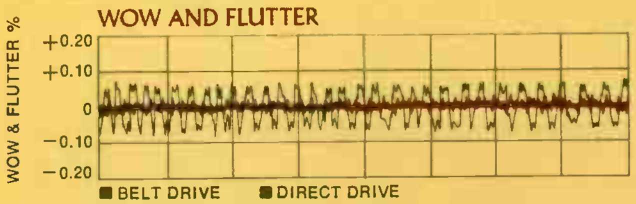
A—33, 45, 78
B—33, 45
C—33 only
D—16, 33, 45, 78
E—16, 33, 45
F—cont. variable

		TURNTABLES										TONE ARMS											
MODEL	MANUFACTURER	Speeds (see letter code)	Wow & Flutter at 33⅓, %	Rumble (NAB) dB	Motor type	Platter diameter, in.	Platter weight, lbs.	Drive	Arm mounting provision	Dimensions, W x D x H, in.	Weight, lbs.	MODEL (for separate arms)	Overall length, in.	Pivot-stylus dist., in.	Vertical bearing	Lateral bearing	Stylus force method	Max. tracking error, deg.	Cartridge weight range, gms.	Arm resonance, Hz	Stylus force range, gms.	Weight, if sep., oz.	Price	SPECIAL FEATURES
REK-O-KUT (CCA)	B12-71	A	0.05	-40	Sync.	12	6	Idler	On order	15¼ x 15¾ x 6	16													
	12-72	A	0.08	-40	Sync.	12	4	Idler	Drld. in base	15¾ x 15¾ x 6	16												194.50	
	CVS-12	F	0.09	-35	Sync.	12	4	Idler	Drld. in base	15 x 16 x 5	15												169.50	
												S-320	12	8¼	Ball	Ball	Bal. & spg.	1.0	7-20	9-12	0-10	24	54.95	
												S-260	16	11	Ball	Ball	Bal. & spg	1.0	7-20	9-12	0-10	28	64.50	
												S410	12	9	Ball	Ball	Bal.	1.25	7-20	10	0-8		69.50	Cueing lever; var. anti-skate.
												102	13	9½	Ball	Ball	Bal.	1.25	7-20	10	0-8		59.95	Calib. stylus force.
SANSUI	SR 1050C	B	0.07	40	Sync.	12	2.9	Belt	Integ.	17¼ x 13¼ x 7½	21½		8¾		Knife	Knife	Bal.	1.5					139.95	Damped cueing; cart. incl.
	SR 2050C	B	0.07	40	Sync.	12	2.9	Belt	Integ.	17½ x 13⅞ x 7½	26		8¾		Knife	Knife	Bal.	1.5					189.95	Damped cueing; cart. incl.
SANYO	TP 80SA	B	0.1	-42	Hys. sync.	11.8	2¼	Belt	Integ.	18½ x 15 x 7¼	22				Ball	Ball	Bal	+2	12 max.	10	½-4		199.95	Auto tonearm p/u & platter shutoff.
SHURE												3009 Imp		9	Knife edge	Ball	Bal.		4-9		0-1.5		135.00	
												3009 S/2		9	Knife edge	Ball	Bal.		4-9		0-1.5		147.50	Detachable shell.
SONY	PS 5520	B	0.1	43	hys sync	12	2¼	Belt	Integ	17¾ x 15½ x 6⅞	18 lb. 11 oz.		11½	8½	Pivot	ball	bal		4-14		0-3		159.50	Auto lead-in; auto ret., repeat.
	PS-2251	B	0.04	58	AC Servo	12	3 lb. 5 oz.	Dir.	Integ	19¼ x 16⅛ x 7¼	33		13½	9⅝	Pivot	ball	bal.		4-17		0-3		349.50	Speed tuning w/built-in stroboscope.
	PS-2251 LA	B	0.04	58	AC Servo	12	3 lb. 5 oz.	Dir.	Mtg. brd.	19¼ x 16⅛ x 7¼	31												299.50	Same as PS-2251.
												PUA-237	13⅜	9⅜	ball	ball				9	0-3		85.00	
SUGDEN (HERVIC)	Connoisseur BD'	B	0.1		Hys.	10¼	2	Belt	Integ.	13¾ x 15½ x 4½	7												129.20	Hyd. cueing: -60 dB rumble RIAA.
												SAUZ	12	9	Ball	Ball	Bal.	0.5		7	0-6		46.70	45/45° gimbal mound; integ. bias.
TECHNICS BY PANASONIC	SP-10	B	0.03	-70*	Direct	12	6	Direct	Sep. base	14 x 14 x 4	20												419.95	Direct drive; strobe; var. pitch; *DIN B, DIN A: -65.
	SL-1100 A	B	0.03	-70*	Direct	13¾	4.4	Direct	Integ.	20⅛ x 15⅝ x 7¾	28			9¼			Bal.	1.75	2-9.5	10	0-5		359.95	Direct-drive; var. pitch; strobe; damp. cueing; *DIN B, DIN A: -65; dust cover incl.
	SL-1200	B	0.03	7*	Direct	13	3.9	Direct	Integ.	16¼ x 13⅞ x 7¼	22			8¾			Bal.	2.0	2-9.5	10	0-4		279.95	Same as above.
THORENS (ELPA)	TD-125AB Mk II	E	0.08	48	sync.	12	8½	Belt	Int.	18 x 14 x 5	32	TP-16	12	9.06	Ball	Ball	Bal. & spg.	0.2	max. 14	8	0-4	10½	275.00	
	TD-125B Mk II	E	0.08	48	syn.	12	8½	Belt	mnt. board	18 x 14 x 5	32												375.00	
	TD-160C	B	0.09	37	syn.	12	8½	Belt	Int.	15⅝ x 12⅞ x 5	16½	TP-16	12	9.06	Ball	Ball	Bal. & spg.	0.2	max. 14	8	0-4	10½	200.00	
V-M	1579	B	0.1		Sync.	11¾	3	Belt	Integ.	17 x 13 x 5	12		12¾	9½	Flex	Cone pt.	Bal. & spg.	1.5	3-9	11	0-4			
YAMAHA	YP-700	B	0.05		Syn.	12	2.6	Belt	Integ.	19 x 16 x 6	20		12	8.7	Ball	Ball	Static	1.5	5-15	8	0-4		239.95	W/cart., base cover; adj. anti-skate; auto-cut/return.

Our most expensive turntable has direct drive. So does our least expensive.

When Technics introduced direct drive, we set new standards for turntable performance. That's why we use direct drive exclusively in all Technics turntables.

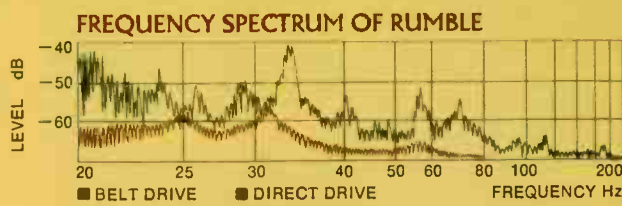
No other system measures up to direct drive. Wow and flutter are less than 0.03% WRMS. And rumble is down to –65dB (DIN A) and –70dB (DIN B).



Our DC motor has no noise- or static-producing brushes and virtually none of the hum normally found in AC motors. It reaches playing speed in half a revolution and has electronic speed control that prevents speed changes due to line fluctuations.

All Technics turntables have illuminated stroboscopes, cast aluminum platters and variable pitch controls. The SL-1200 also has a precision tone arm, viscous damped cueing and low-capacitance 4-channel phono cables.

So does the SL-1100A but with a heavier platter, bigger motor and longer tone arm.



And the SP-10 is for those who insist on choosing their own tone arm.

No matter which Technics turntable you choose, you get the finest drive system that money can buy. But don't take our word for it. Read the reviews.* And you'll agree.

The concept is simple. The execution is precise. The performance is outstanding. The name is Technics.

*SP-10: Audio, 8/71; Stereo Review, 9/71.
SL-1100A: Stereo Review, 7/73; High Fidelity, 9/73.

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Technics
by Panasonic

automatic or manual turntables which one to buy?

Joe Lesly*

As soon as you decide you'd like to buy components, the next step probably depends on whether you know someone who's knowledgeable about high fidelity. If you do, the rest is easy. You just follow the advice that's offered. But if you don't, and try to learn by yourself, you must be prepared to learn more than you want to know. Just flip through some of the pages in this particular issue, for example, and see how many ways of playing records there are. That is, how many makes and models of record playing equipment are available. All you really want to know is: Which one best suits my purpose?

As the writer is associated with one of the manufacturers, it would be easy for us to tell you. But that wouldn't be fair to you or the editors. Nor would it be as interesting a challenge to us as the task we've undertaken: to be as objective as our human frailty will allow.

Now, back to your question: Which one to buy? To decide that, you should know what to look for. And to know what to look for, you should pause to consider some fundamentals, obvious as they may be, about what happens from the time you place a record on the platter and the time you return it to its jacket.

After a record is placed on the platter, the next thing that happens is the movement of the tonearm from its resting post to the lead-in groove. What you have to decide is whether you want the tonearm to move by hand or whether you prefer to move a switch and let it all happen automatically.

That choice brings us all the way back to the tired old subject of record changer vs. manual turntable. We won't get into that one again, other than to point out that this question has long since become one of convenience, not quality.

Actually, it's more than a simple question of convenience. If you've recently handled a tonearm set to track at one gram, you may well have felt uneasy about it. One gram feels like no weight at all, and it's no easy matter to handle it with confidence. What's more, one-gram tracking means there's a very fragile suspension between the body of the cartridge and the tiny diamond tip at the end.

So even if you don't really need a "changer" per se, you may well need the security of an automatic start. (And stop, for that matter.) Fortunately, most manual turntables are provided with a "cueing" device that takes care of the set-down and lift-off function, leaving you only the job of moving the tonearm to and from the record. Plus, of course, remembering to set the cueing lever for the action needed. And finally, hoping that anyone else in your family who uses your system will also remember, and use it properly.

OK. One way or another, the tonearm has descended to the rotating record, and the diamond stylus is now in the groove. You can now walk away and enjoy the music for the next twenty minutes or so. But to be sure you're enjoying the same music every time you play the same record, you want to be sure that the stylus and the record are getting along together just fine.

You know, of course, that the diamond stylus is the hardest substance on earth, and that the vinyl record is anything but. In fact, a crude but homey comparison might be something like dragging a rounded end of a stick along the surface of a bowl of jello. The parallel object of that little example would be to avoid breaking the thin skin of the jello.

In principle, what you hope will happen to your record is: nothing. The stylus will move this way and that way, very obligingly, as the contours of the record groove direct. And, of course, their "directions" occur at a rate of something up to fifteen thousand a second.

For "nothing" to happen, the tonearm holding your cartridge should 1) be precisely balanced, 2) have the correct stylus pressure applied to meet the stylus compliance requirements and 3) apply that pressure equally on both walls of the stereo groove. (Not every one agrees with the importance of that last criterion, which refers to anti-skating, but their ranks have noticeably thinned as more has become known about the need for balanced tracking.)

Just one other thought for you about the actual action of a record in play. Quickly now, what makes the tonearm move across the record from lead-in groove to run-out groove? Did the answer come to you right away? Many people have never given it any thought. Answer: pressure of the outer wall of the record groove as the groove spirals inward.

Another quick question: What resists that inward movement of the tonearm? Answer: the friction of the bearings in the tonearm pivots. When you consider that the stylus might be riding in the groove with perhaps one gram of pressure, it does not require much friction in the pivot to unbalance the tracking on the two walls and even cause occasional skipping. And worse, permanent damage to the recorded material in the groove.

If all this seems academic to you and more than you want to know, take a moment right now and add up the money you've already spent on records, and the money you're likely to spend

*Vice president, Ries, Cappiello, Colwell, Inc. (*agents for Dual*) in the future. Chances are you've already spent more on records than on all your other components combined. Assuming we're in agreement on that score, we can now turn to the question of evaluating the record playing equipment available at the typical audio dealer.

Unlike receivers and speakers, whose insides are effectively sealed, record players are accessible to the eye and hand. Most elements of their design, materials and workmanship are exposed, either above or below.

As you look up and down the shelves, you'll see a sampling of the products described in this magazine, ranging from very low-priced to fairly high-priced record changers and automatic and manual turntables. It's fair and accurate to say that price and quality go together, and we think it's clear that we hope you will put quality first and not short-change yourself or your records with false economy. After all, damage or premature wear on a record is permanent.

With the salesman's permission, you should operate each prospective turntable a few times, using whatever start and stop switches are provided, and using the cueing device on those units which have them. Note how smoothly and quietly everything works (or doesn't).

Now pay attention to the various tonearm settings: for balance, tracking pressure and anti-skating. Again, if the salesman allows, you should place all the settings back to neutral and unbalance the tonearm. Then start all over again, just as you would in your own home. Balance the tonearm with the weight of the cartridge. Apply tracking force, then anti-skating. Just performing these functions will give you a sense of how well the machine is made.

You might now pay attention to some of the features and refinements that each model offers. Although the "experts" might disagree on the importance of each, the more seriously minded turntable manufacturers prefer to include refinements that they consider necessary. For example, some models have two settings for anti-skating: one for conical styli, the other for elliptical styli. This refinement recognizes the fact that the narrower tracing edge of the elliptical stylus penetrates slightly deeper into the groove wall, resulting in more frictional pull and hence more skating. So the "elliptical" calibration applies a bit more anti-skating than the equivalent "conical" calibration.

One refinement the better record changers have adopted from single-play units is a single-play spindle that fits snugly into the platter and rotates with it. This type of spindle centers the record more precisely and helps reduce wow and flutter. Also, it certainly makes good sense for a spindle to rotate with the record instead of presenting potential resistance which can enlarge center holes and increase eccentricity.

When you operate the cueing lever, pay particular attention to the way the tonearm responds. The better cueing systems are damped, so the tonearm floats down at a constant slow rate of speed no matter how fast you move the lever. And the best systems also damp the tonearm on the way up, so it won't bounce. If you don't think you will really use the cueing feature much, consider the problem when you're listening to a record when the phone rings. Without cueing, you have the choice of turning the record off or paying little attention to it. With cueing, you can lift the tonearm off the record, then resume where you left off when the call is over.

So much for the tonearm. Now for the platter and what makes it go around. Lower priced models have a stamped steel platter. Higher priced ones have nonferrous cast platters. The highest-priced ones also have cast platters which are dynamically balanced. In addition, all things being equal, the heavier the platter the better its flywheel action in smoothing out the fluctuations of the motor itself.

Any acceptable record changer has a four-pole motor, as the two-pole motor belongs to the "toy" category. But there are differences among four-pole motors. The better ones are able to resist line voltage changes and thus maintain speed constancy, which is the special virtue of the synchronous motor. The best of the conventional motors is the type which is both inductive and synchronous. These motors use their induction elements to provide the torque required to get the platter up to speed quickly, then the synchronous element takes over to lock the speed into the 60H frequency of the AC power line.

A new kind of motor has recently appeared in a very few of the highest-priced single-play turntables. This motor is electronic and is energized by a regulated DC power supply. Its particular advantage is its low rotational speed which reduces vibration (hence rumble). Even electronic motors differ from each other, as the best of them not only rotate at low speed, but at the exact speed required for the record, 33⅓ or 45 rpm. Thus, these motors don't require any speed-reduction system, such as friction-idler or rim-belt, and can drive the platter directly. Be forewarned that such motors come with turntables priced at over $300.

One useful feature associated with motors and drive systems is a variable speed control. Since records are made to rotate at certain fixed speeds, whether 33⅓ or 45 rpm, you may wonder why the platter shouldn't always rotate at these speeds and let it go at that. For one thing, your taste may lead you to prefer a certain record that's pitched slightly higher or slightly lower. If you ever want to play a live instrument along with a record, you'll find it easier to tune the record to a piano, for example, than vice versa. And camera buffs also find this feature useful in timing a recording to a length of film.

In case you haven't noticed the absence of any comment about wow, flutter and rumble, we'll mention them right now. All record players have some wow and flutter measurements below 0.1%. And rumble (weighted) should be below 50 dB. But it will take a very good ear to detect either one unless they are grossly evident, and any record player good enough to be submitted for published test reports, such as in Audio, will be adequately quiet.

At this point, we hope we haven't told you more about record players than you really want to know. And if you remember nothing else, please remember that only one component handles your precious records. You want that component to be a very good one. Æ

Automatic Turntables

B&O Beogram 3000

Dual 1216

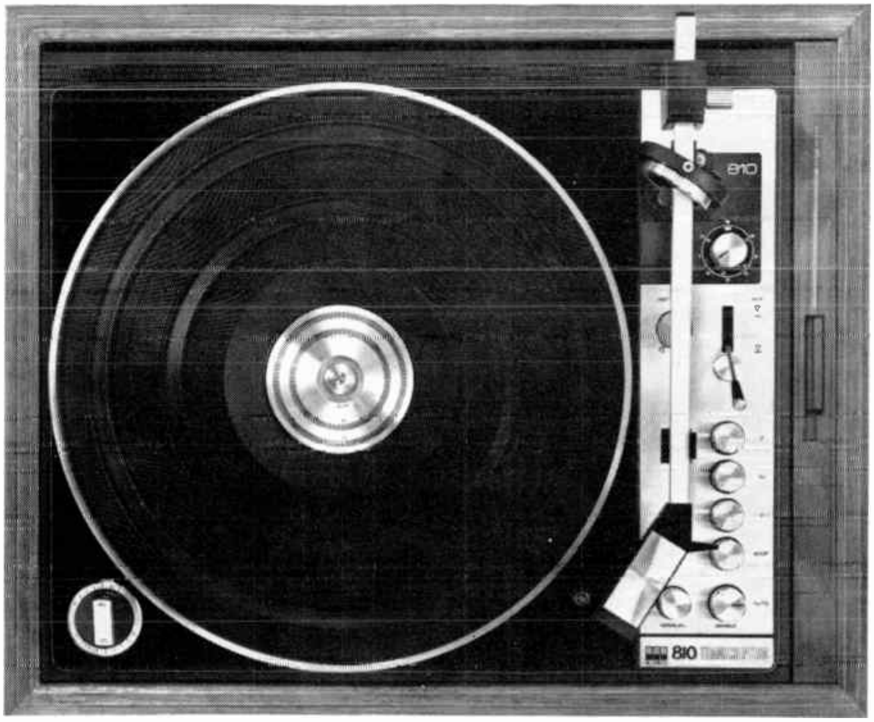
Elac 50HII

BSR-McDonald 810

Speeds (use letter code)

A - 33, 45, 78
B - 33, 45
C - 33 only
D - 16, 33, 45, 78
E - 16, 33, 45
F - Cont. variable

MANUFACTURER	MODEL	Speeds (see letter code)	Platter diameter, in.	Wow and flutter at 33⅓, %	Rumble (NAB) dB	Max. tracking error, deg.	Pivot-stylus dist., in.	Arm type	Tracking force range	Arm resonance, Hz	Max. stack records	Change cycle at 33⅓, secs	Clearance below board, in.	Clearance above board, in.	Overall W x D, in.	Overall height, in.	Weight, lbs.	Price	SPECIAL FEATURES
B & O	3000	B	12	0.075	37**	0.126	9	Bal. & spg.	0-3.5	13	0	5	*	*	17⅜ x 13	4¾	20	265.00	*Auto/integ. sys.; incl. base, cover & SP-12A cart. ($85.00. **DIN
BSR	810/X	B	12	*	−55	0.5	8.562	Cntr. wt. ball brg.	0-4	7	6	14	3	4½	17⅛ x 15	9¼	22	239.45	*Wow less than 0.1%, flutter less than 0.05%; seq. cam sys.; dual volt. motor.
	710/X	B	12	*	−55	0.5	8.562	Cntr. wt. ball brg.	0-4	7	6	14	3	4½	17⅜ x 14⅝	8¾	19	204.80	Same as above.
	610A/X	A	11	0.26	−30	4.5	7.476	Cntr. wt.	0-6	11	6	8	3	4	15¼ x 14⅛	7¼	17	130.45	
	510A/X	A	11	0.26	−30	4.5	7.476	Cntr. wt.	0-6	11	6	8	3	4	15¼ x 14⅛	7¼	14	105.45	
DUAL (UNITED AUDIO)	1229	A	12	0.04	−55	0.3	8¾	Bal. & spg.	1-12	8-14	6	13	3	5	14¾ x 12	8	19	249.95	Gimbal susp.; adjust. vert. track angle; 6% pitch contl. sync. mtr.; cueing; ill. strobe.
	1218	A	10⅝	0.06	−45	0.5	8¼	Bal. & spg.	1-12	8-14	6	11	2⅝	5	13 x 10¾	7⅝	14	189.95	As above less strobe.
	1216	A	10⅝	0.08	−45	0.5	8¼	Bal. & spg.	1-8	8-14	6	11	2⅝	5	13 x 10¾	7⅝	13	149.95	Pitch contl.; damped cueing; sep. anti-skate for con. & ellip. stylii; diecast platter.
	1214	A	10⅝	0.08	−45	0.5	8¼	Bal. & spg.	1-8	8-14	6	11	2⅝	5	13 x 10¾	7⅝	13	119.95	Pitch contl.; damped cueing; built-in antiskate.
ELAC (BENJAMIN)	50H Mk II	A	12	0.06/ 0.02	−40	0.4°	7¼	Bal. & Spg.	.5-6.5	8	10	10	3¾	5¾	14 x 12½	10	18	225.00	Hys. syn. mtr.; pitch adj.; ill. dig. strobe; sty. overhang adj.
	760	A	12	0.06/ 0.02	−44	0.4°	7¼	Bal. & Spg.	.5-6.5	8	10	10	2⅞	5½	14 x 12	10	18¼	179.00	Pitch adj.
	650	D	10⅜	0.07/ 0.03	−38	0.5°	7⅛	Bal. & Spg.	0.5-6.5	10	10	12	2⅝	5⅝	13½ x 11½	9	17	149.50	
	625	D	10⅜	0.07/ 0.03	−38	0.5°	7⅛	Bal.		10	10	12	2⅝	5⅝	13½ x 11½	9	15	119.95	



We gave our best turntables the shaft.

And they're better because of it.

The BSR 810 and 710 have their brains in their shaft. A carefully machined metal rod holding eight precision-molded cams. When the cam shaft turns, the cams make things happen. A lock is released, an arm raises and swings, a record drops, a platter starts spinning, the arm is lowered, the arm stops, the arm raises again, it swings back, another record is dropped onto the platter, the arm is lowered again, and so on, for as many hours as you like.

Deluxe turntables from other companies do much the same thing, but they use many more parts—scads of separate swinging arms, gears, plates, and springs—in an arrangement that is not nearly as mechanically elegant, or as quiet or reliable; that produces considerably more vibration, and is much more susceptible to mechanical shock than the BSR sequential cam shaft system.

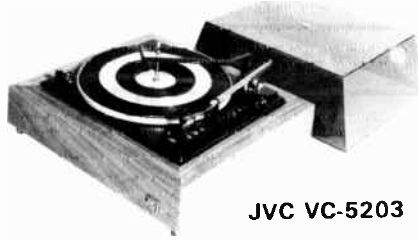
When you buy a turntable, make sure you get the shaft. The BSR 710 and 810. From the world's largest manufacturer of automatic turntables.

BSR

BSR (USA) Ltd., Blauvelt, New York 10913

Automatic Turntables

Garrard Zero 100



JVC VC-5203

Radio Shack 45

Glenburn

Speeds (use letter code)

A - 33, 45, 78	D - 16, 33, 45, 78
B - 33, 45	E - 16, 33, 45
C - 33 only	F - Cont. variable

MANUFACTURER	MODEL	Speeds (see letter code)	Platter diameter, in.	Wow and flutter at 33⅓, %	Rumble (NAB) dB	Max. tracking error, deg.	Pivot-stylus dist., in.	Arm type	Tracking force range	Arm resonance, Hz	Max. stack records	Change cycle at 33⅓, secs.	Clearance below board, in.	Clearance above board, in.	Overall W x D, in.	Overall height, in.	Weight, lbs.	Price	SPECIAL FEATURES
GARRARD (BIC)	Zero 100C	B	11½"	0.06/ 0.025	−51	0	7½	Bal. & Wt.	0-3	8	6	10	3	4¾	15¼ x 14⅛	7⅜	14½	209.95	Zero tang. track. arm, var. speed, ill. strobe, mag. anti-skate visc. damp. cue., 15° adj., overhang adj.
	Zero 92	A	11½"	0.07/ 0.025	−48	0	7½	Bal. & Wt.	0-4	8	6	10	3	4¾	15¼ x 14⅛	7⅜	14½	169.95	Zero tang. track. arm, slide w. anti-skate, visc. damp. cue., 15° adj, overhang adj.
	Model 82	A	10½"	0.08/ 0.025	−48	0.75	8	Bal. & Wt.	0-4	8	6	10	3	4¾	15½ x 14⅛	7⅜	14½	119.95	Visc. damp cue., 15° adj., overhang adj., slide w. anti-skate & stylus pressure.
	Model 70	A	10½"	0.09/ 0.025	−45	0.85	7¼	Bal. & Spring	0-5	10	6	10	3	4½	13 13/16 x 12⅜	7	10½	89.95	2 p. record drop, damp. cue., anti-skate, built-in stylus pressure gauge, CB arm, Syn. motor.
	Model 62	A	10½"	0.1/ 0.03	−44	0.85	7¼	Bal. & Spring	0-10	10	6	10	3	4½	13 14/16 x 12⅜	7	10½	69.95	2 p. record drop, damped cueing, anti-skate.
GLENBURN/ McDONALD	2110	D	11	0.2	−39	2.0	7½	Spg.	1.5-6	5	8	10	2	4	13⅛ x 11¼	6	8	74.00	Incl. base, dust cover & Shure M75 cart.; cueing: fact. adj. anti-skate.
JVC	VC-5203	D	11	0.1	−45		8	Bal. & Spg.			6	10			16⅜ x 14¼	7¼	13⅛	89.95	With mag. cart., base, dust cover.
	4VC-5244 4 chan.	B	11	0.1	−45		8	Bal. & Spg.			6	10			15¾ x 17¼	7½	19	249.95	CD-4 4-chan. demodulator built-in; base, cover, mag. cart. w/Shibata stylus.
MAGNAVOX	1K8821	A	11	0.28*	−43*	5.0	7½	Bal. & Spg.	0.5	5	6	10			16½ x 14½	7¾	15	89.95	*DIN
PE (IMPRO)	3060	A	10⅝	0.08	59	0.5	8¼	Bal. & Spg.	1-10		6	13	2½	4	13 x 10½	6½	10½	195.00	Sep. anti-skate for con. & ellip.; sync. mtr.; gimbal arm; track angle adjust.
	3015	A	10⅝	0.12	58	0.5	8¼	Bal. & Spg.	3-15		6	13	2½	4	13 x 10½	6½	9½	149.95	Fail safe stylus contl.; pitch contl.; rotat. single play spindle; damp. cueing; anti-skate.
	3012	A	10⅝	0.15	56	0.5	8¼	Spg.	3-15		6	13	2½	4	13 x 10½	6½	9½	109.95	As above less anti-skate.
RADIO SHACK	45	D	11⅞	0.1	−50	0.6	7¼	Bal. & Spg.	¾-6	2	6	12			16¾ x 14¾	4¼	20	164.95	W base, $33.95 mag. cart.
	40C	D	11⅞	0.1	−50	0.6	7¼	Bal. & Spg.	1-6	2	6	12			16¾ x 14¼	4¼	20	134.50	W. base, $22.95 mag. cart
	LAB-36A	A	11	0.18	−29			Bal. & Spg.	2 min.		6				15¼ x 13¼	6½	10½	79.95	W. base, $22.95 mag. cart.
	LAB-24C	A	11	0.18	−29			Bal. & Spg.	2 min.		6				15¼ x 13¼	6½	8	64.50	W. base, $17.95 mag. cart.
V-M	1670	D	11	0.28/ 0.14					2½-3½	5					15⅛ x 14⅛	7¾	16	79.95	Damp.; Shure mag. cart w/dia. stylus; pop filt.; muting swit.; cover; base; auto off.

For those content to settle for mere greatness in an automatic.

In order to tell you about the merely great Miracord 760 automatic turntable, we must admit that we make a slightly greater model, the Miracord 50H Mark II. But to call the 760 "second best" is to call a Bentley "just another car."

But before we tell you about the differences, we'll tell you about the remarkable similarities.

Both turntables begin with ELAC's unique, no-shake push-button control system. This takes the jolt and jar out of operation and reduces the chance of record damage.

The 760 tracks with dead accuracy as low as ½ gram stylus pressure, and the anti-skating device is precise beyond belief.

Pitch control? Of course. Like our top-of-the-line Mark II, the 760 allows you to vary speed over a 6% range (equal to a semi-tone in pitch). And a built-in stroboscope allows for simple, unerring speed adjustment.

The 760's 12," one piece, die-cast turntable platter is dynamically balanced for smooth performance and consistent speed.

So what's the difference between the Miracord 760 and the 50H Mark II (which costs $35 more)? It's merely in the motor. The 760 has a specially designed spectacularly consistent asynchronous motor, and next to 50H Mark II its speed accuracy is virtually unsurpassed in the audio field.

If you're looking for the ultimate in fidelity, you'll want a handcrafted turntable by ELAC. And we invite you to write us for literature on the entire ELAC line.

Just a word of caution. Because you can't rush craftsmanship, you may not find the 760 or the 50H Mark II readily available. But we'd rather be great than easy to get. If you find yourself shopping around for our turntable, take comfort in this obvious fact: you don't find a Rolls Royce dealer on every corner. Because greatness can't be mass produced.

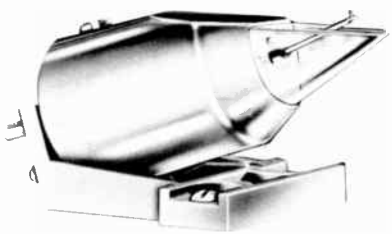
ELAC Products/Benjamin Electronic Sound Company, Farmingdale, N.Y. 11735.

MIRACORD 760

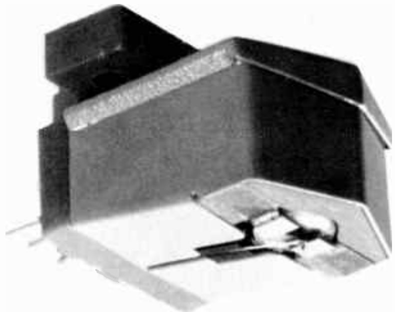
ELAC

You can't rush craftsmanship.

Phono Cartridges



B&O



Decca London Export

Elac STS-444

Stylus Type
C - Conical
E - Elliptical

MANUFACTURER	MODEL	Frequency response, Hz ± 2 dB.	Separation, 1000 Hz, dB	Separation, 10 kHz, dB	Output, mV/cm/sec.	Tracking-force range, gms.	Load resistance, ohms	Stylus type (see letter code)	Stylus radius (radii) mils	Replacement	Weight, gms.	Price	Special Features
ASTATIC	181d	60-12k ± 6	25	18	31	2-3	143k	C	0.7 x 3.0	user	1.7	(14.50	Piezoelectric ceramic. Nominal Response relative to RIAA. No electrical equalization required.
	157d	90-14k ± 6	24	14	33	3-4.5	143k	C	0.7 x 3.0	user	3.4	$13.50	Same as above.
	155d	70-11k ± 6	23	11	125	3.5-5	1 mag	C	0.7 x 3.0	user	3.4	$11.95	Same as above.
	139d	90-13k ± 6	24	14	78	4.5-6	500k	C	0.7 x 3.0	user	3.4	$13.75	Same as above.
AUDIO DYNAMICS	XLM	10-25k	30	30	4	0-6	47k	E	0.3 x 0.7	User	3.8	50.00	IM = 0.3% @ 14.2 cmS.
	VLM	10-22k	30	30	4	1-1.4	47k	E	0.3 x 0.7	User	3.8	46.00	
	20XE	10-18k	20	20	6	1-2½	47k	E	0.3 x 0.7	User		22.00	
	Q-36	10-22k ± 3	25	25	4.5	¾-1½	47k	E	0.3-0.7	User		45.00	
	Q-32	10-20k ± 3	30	30	4.5	1-2	47k	E	0.3-0.7	User		36.00	
	Q-30	10-20k ± 3	30	30	4.5	1-2	47k	C	0.5	User		27.00	
AUDIO-TECHNICA	AT20SL	5-50k	25		2.7	1¼-2		E*	*			150.00	*Shibata configuration
	AT15S	5-45k	25		2.7	1¼-2		E*	*			100 00	*Shibata configuration.
	AT14S	5-45k	25		2.7	1¼-2		E*	*			75.00	*Shibata configuration.
	AT13E	10-45k	25		3.5	1-2		E	0.2 x 0.7			64 95	
	AT12E	15-30k	22		3.5	1¼-2		E	0.4 x 0.7			54.95	
	AT11E	15-28K	20		4.8	2-3		E	0.4 x 0.7			44.95	
	AT11	15-25K	20		4.8	2-3		C	0.7			34.95	
	AT10	20-20k	20		4 8	2¼-4		C	0.7			24.95	
B & O	SP-12	15-25k ± 3	25	20	1.0	1-1½	47k	E	0.2 x 0.7	User	8 5	85.00	Naked diamond stylus; pat. moving micro-cross constr
	SP-14	20-16k ± 2.5	20	16	1.0	1½-2½	47k	C	0.6	User	8.5	45.00	Naked diamond stylus; pat. moving micro-cross constr.
	SP-10	15-25k ± 3	25	20	1.0	1-1½	47k	C	0.6	User	8.5	75.00	Naked diamond stylus; pat. moving micro-cross constr
DECCA (PAOLI)	London Export Mk V	30-18k	25	18	1 5	2.5-3	50k	C	0 5-0.6	fty	5	135 00	W/resp. curve; pos. scan., no cantil.; <1.0 mgm tip mass; hand polished
	London Mk V	40-16k	20	18	1 5	2 5 3	50k	C	0 5	fty	5	109 50	Pos scan no cantil: <1.0 mgm tip mass; hand polished
	78C	40-16k			1 5	3 5-4	50k	C	2 5	fty	14	35 00	Same as above exc 78 rpm discs
ELAC (BENJAMIN)	STS 244-17	20-20k	22		1 5	1 5-3	47k	C	0 7	user	6 5	25 00	
	STS 344-E	20-22k	24		1 0	1 2	47k	E	0 2 x 0 8	user	6 5	60 00	
	STS 444-E	10-24k	26	17	1 0	¾ 1 5	47k	E	0 2 x 0 8	user	6 5	80 00	

A brief discussion of the Shibata Stylus.

The sole job of a phonograph stylus is to accurately trace the tiny modulations of the groove in a phonograph record. And the task of a phonograph cartridge is to translate these stylus motions into an electrical signal that is an exact duplicate of the signal used to create the groove modulations.

Response to 45,000 Hz.
In the past, this cartridge "team" had to concern itself with a range of signals that did not exceed the limits of hearing, roughly from 20 to 20,000 Hz. With the introduction of 'discrete' 4-channel discs, however, the high frequency capabilities of the system have now been called upon to respond accurately to tones as high as 45-50 kHz. To do this consistently, a new stylus design was required.

Smaller tip needed.
Spherical tips and elliptical styli, designed for mono and 2-channel stereo records, are simply too large and gently rounded to fit the tiny modulations required for the discrete record. A tip with a smaller radius was needed. However, if only the radius at the side of the stylus was reduced, leaving all else the same, the area of the stylus contacting the record would be very small. Thus, if the force of the cartridge on the record was unchanged, then the pressure on the much smaller contact area of the record would rise to unacceptable levels.

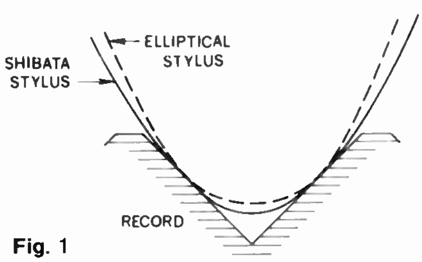


Fig. 1

Contact area increased.
An entirely new stylus shape was needed, and the Shibata stylus is the result. Its most important difference can best be seen by viewing both an elliptical and a Shibata stylus from the front (Fig.1). Note that an elliptical stylus has an essentially rounded tip that contacts the sides of the grooves at just one small point. Pressure at this point is extremely high. The Shibata tip is shaped to better conform to the V-shaped groove, and contact area between record and stylus is increased about 4 times over an elliptical stylus, greatly reducing pressure on the record. Behind this long contact edge the Shibata stylus is cut away in a shallow "V" to permit a very small radius at the contact edge (Fig. 2). It is this small radius that permits proper tracing of frequencies as high as 45,000 Hz.

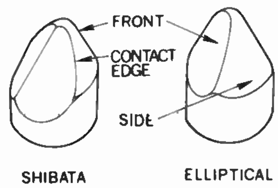


Fig. 2

Reduced record pressure.
In addition to permitting proper operation of discrete 4-channel records, the Shibata stylus offers benefits for every kind of record. By increasing the contact area and reducing pressure at any given point on the record, the Shibata stylus increases record and stylus life by up to four times. Even at very low stylus pressures, most styli will actually deform the record surface momentarily as the groove moves past (Fig. 3). The increased bearing surface of a Shibata tip sharply reduces this effect, especially important if a record is replayed immediately (Fig. 4). And this pressure reduction is absolutely vital for 4-channel records, whose delicate high frequency modulations can actually be wiped off by too much pressure from a conventional stylus/cartridge combination.

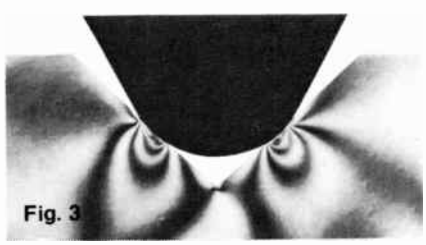


Fig. 3

Stress analysis photo shows elliptical stylus contacts groove in concentrated area with high pressure.

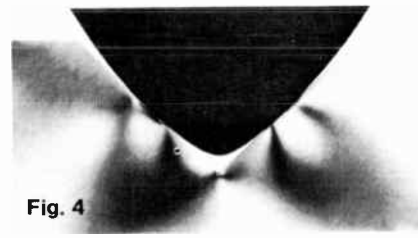


Fig. 4

Shibata stylus contacts groove over larger area, stress lines are reduced, pressure is lower, and groove is less distorted.

Extended high frequency response.
Another benefit of reduced pressure is to raise the effective stiffness of the record, thus raising the resonant frequency of the record/stylus/cartridge system. This extends high frequency response and reduces peaks in the audible spectrum and beyond. This benefit can be heard with all records, especially at the inner grooves of a record where groove velocity is lowest, yet modulation is typically high as music comes to its climax. The Shibata stylus is better able to handle this difficult reproduction problem.

Two basic benefits.
The importance of the true Shibata shape is thus two-fold. It can operate at reasonable stylus pressures, while reducing the stress on any given part of a record for extended record and stylus life. And it permits extension of high frequency response to the limits required by discrete four-channel records.

Four models offered.
Audio-Technica currently offers the Shibata stylus on 4 models of their Dual-Magnet phono cartridges: the inexpensive AT12S, the popularly-priced AT14S, the deluxe AT15S, and the limited edition AT20SL. All can be used in a broad range of tone arms and automatic turntables to properly reproduce every kind of modern stereo, matrix four-channel, and discrete four-channel recordings.

audio-technica®

AUDIO-TECHNICA U.S., INC., Dept. 144A, 1655 W. Market Street, Fairlawn, Ohio 44313

Phono Cartridges

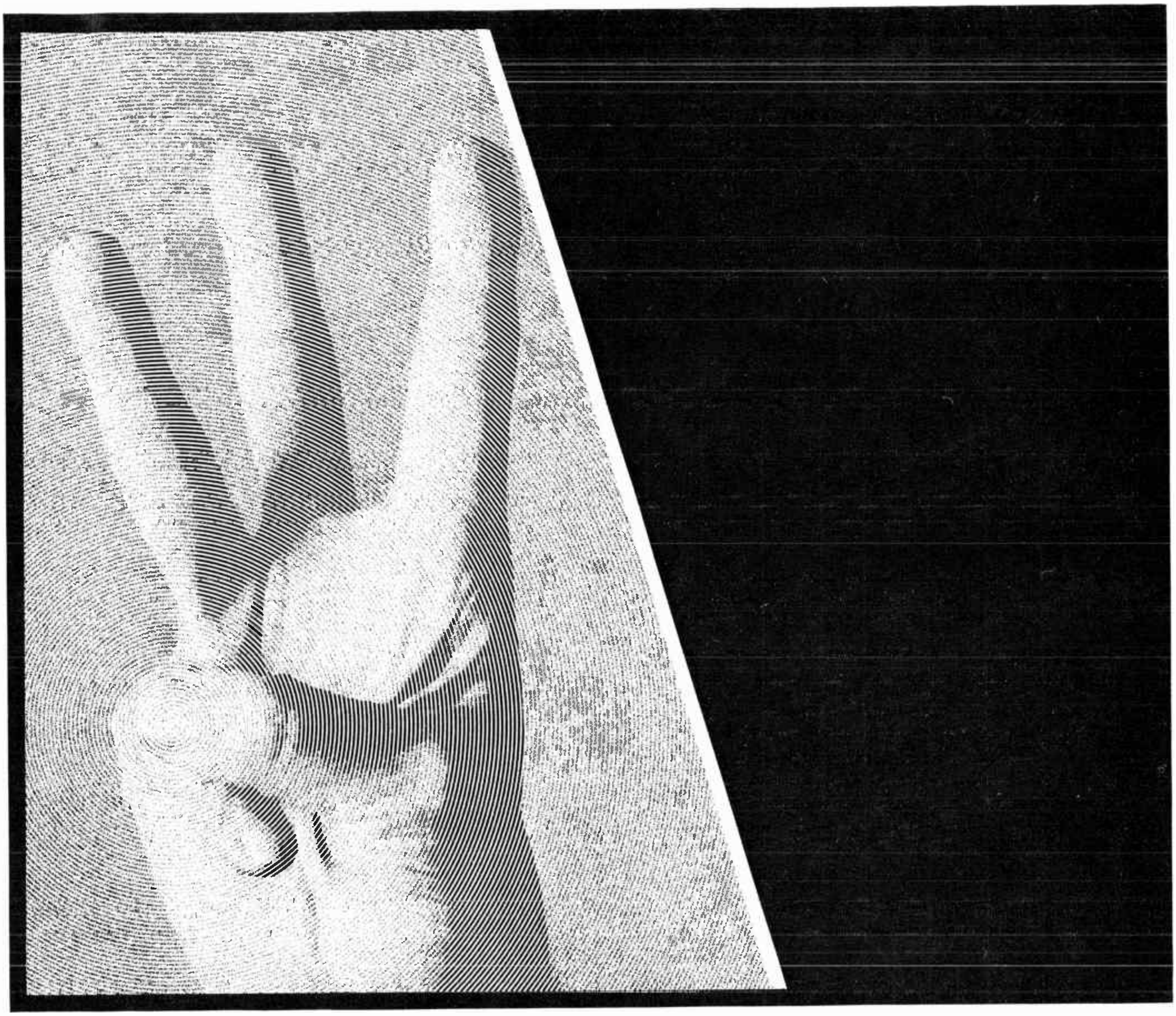
Empire 1000 ZE/X

JVC 4MD-20X

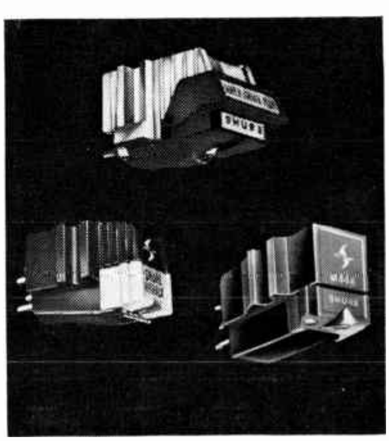
Shure V-15 III

Stylus Type
C · Conical
E · Elliptical

MANUFACTURER	MODEL	Frequency response, Hz ± 2 dB.	Separation, 1000 Hz, dB	Separation, 10 kHz, dB	Output, mV/cm/sec.	Tracking-force range, gms.	Load resistance, ohms	Stylus type (see letter code)	Stylus radius (radii) mils	Replacement	Weight, gms.	Price	Special Features
EMPIRE	1000 ZE/X	4-40k	35	25	1 2	¼-1¼	47k	E	0 2 x 0 7	User	7	99 95	
	999 VE/X	6-36k	35	25	1.2	¼-1¼	47k	E	0.2 x 0.7	User	7	79.95	
	999 TE/X	8 36k	35	25	1 2	½-1½	47k	E	0 2 x 0 7	User	7	64 95	999SE/X similar but freq resp of 8 32k $49.95
	999E/X	10-30k	35	25	1 5	¾-1½	47k	E	0 2 x 0 7	User	7	34.95	909E/X same but resp of 12 25k $29 95
	90EE/X	15 25k	35	25	1.5	¾ 2	47k	E	0 2 x 0 7	User	7	29.95	
	66PE/X	8-30k	35	25	1 8	½-2	47k	E	0 2 x 0 7	User	6	44 95	66E/X sim but resp is 10-30k: radii. 0 3 x 0.7 $29 95: 66/X sim but resp 10-25k trkg. rng ¾-2 & conical (0 7 mils) stylus $19 95
	4000 D III	6-60k	35	35*	1 0	½-1½	47k	4 ch	0 1**	User	5	149 95	Designed for 4-chan : *rear 30dB: **4 dimens
	4000 D II	6-60k	35	35*	1 0	½-1½	47k	4-ch.	0 1**	User	5	124.95	Designed for 4-chan : *rear 30dB: **4 dimens
	4000 D I	6 50k	35	35*	1 0	¾-2	47k	4 ch	0 1**	User	5	84 95	Designed for 4-chan : *rear 30dB: **4 dimens
JVC	4MD-20X	20-60k	30	25	2 0	1 5-2	47k	*		User	8	69 95	*Shibata stylus. for use w/4-chan discs
OLSON	PC-195	15-25k	30		2.5	1 5 4	47k	E	0 4 x 0.7	User	14	24 98	
PICKERING	UV 2400Q	10-50k	35		0 7	1 3		*		User	5	124 95	*Quadrahedral for CD-4: Oustamatic brush
	XV15/1200E	10 30k	35	25	0 8	½-1¼	47k	E	0 2 x 0 7	User	5	79 95	
	XV15/750E	10-25k	35	25	0.8	½-1½	47k	E	0.3 x 0.7	User	5	65.00	
	XV15/400E	10 25k	35	25	1 0	1-2	47k	E	0 4 x 0 7	User	5	54 95	
	XV15/350	10-25k	35	25	1 1	1-3	47k	C	0 7	User	5	39 95	
	XV15/200E	10-25k	35	18	1 4	2-4	47k	E	0.4 x 0 7	User	5	49 95	XV15/150 sim. but 0.7 mil con. stylus, $34.95.
	XV15/140E	10 20k	35	16	1 4	3-5	47k	E	0 5 x 0 7	User	5	34 95	XV15/100 sim. but 0.7 mil con. stylus, $29.95.
	V-15 Micro IV AME	20-20k	30	24	1.0	1.2	47k	E	0.4 x 0.7	User	5	49.95	V-15 Micro IV AM sim. but 1.1 mV output: 0.7 con. stylus, $34.95.
	V-15 Micro IV ATE	20-18k	28	15	1.2	2-4	47k	E	0.4 x 0.7	User	5	39.95	V-15 Micro IV AT sim. but 1.4 mV output: 0.7 mil. stylus, $29.95.
	V-15 Micro IV ACE	20-17k	26	14	1.4	3-5	47k	E	0.5 x 0.7	User	5	29.95	V-15 Micro IV AC sim. but 0.7 mil. stylus, $24.95.
QRK (CCA).	F3	20-20k	25	25	4.0	1.5-3.5	47k	C	0.6	User	5.5	29.95	
RADIO SHACK	R700E	10 25k	25		6.2	¾-1½		E	0 2 0.7	User		33 95	
	R27E	20-20k	25		6.2	1-1½		E	0 2 x 0 7	User		22 95	
	R47EB	20 20k	20		6 2	1½ 3		E	0 4-0 7	User		17 95	
	R25EC	20 20k	25		6 2	2 5-5		E	0 4-0 7	User		15 95	
SHURE	V15 III	10-25k	28	20	3 5	¾-1¼	47k	E	0 7 x 0 2		6	72 50	
	M91ED	20 20k	25		5 0	¾-1½	47k	E	0 7 x 0 2		5.5	54 95	M75ED Type 2 similar but with stylus guard.
	M91E	20 20k	25		5 0	¾-1½	47k	E	0 7 x 0 2		6	49 95	
	M93E	20 20k	25		6 2	1½-5	47k	E	0 4 x 0 7		6	39 95	
	M75G Type 2	20-20k	25		5 0	¾-1½	47k	C	0 6		6	38 45	
STANTON	780 4DQ	10 50k	35		0 7	1-3		*		User	5	125 00	*Quadrahedral for CD 4: "Longhair brush"
	600A	20 20k	35		1 0	2-4	47k	C	0 7	User	5	45 00	
	600EE	20-20k	35		1 0	1 2	47k	E	0 3 x 0 7	User	5	55 00	600E sim. but trkg. force rng. 1½ 3 gms., $50.00.
	681SE	10-20k · 1	35	26	1 1	1½-3	47k	E	0 4 x 0 7	User	5	66 00	681A sim. but 0.7 con. stylus, $66.00.
	681EE	10 20k · 1	35	26	0 8	¾ 1½	47k	E	0 2 x 0 7	User	5	72 00	
	500E	20 20k	35	22	1 0	2 5	47k	E	0 4 x 0 7	User	5	35 00	500A sim but 0 7 mil con stylus $30 00
	500EE	20-20k	35	22	1 0	1 2	47k	E	0 3 x 0 7	User	5	40 00	500AA sim but 1 2½ trkg frc . 0 5 con stylus $35 00



Best. Best. Best.



Permit us this momentary bit of self-indulgence, because our intentions are pure: to assist you in choosing the best phono cartridge for your hi-fi system, within the practical limitations of your audio budget. To begin, if you feel uncomfortable with anything less than state-of-the-art playback perfection, we heartily recommend the Shure V-15 Type III, a cartridge of such flawless performance it is the perfect companion to the finest turntables and tone arms available today — and those coming tomorrow. At a more moderate level of performance and price, we suggest the Shure M91ED, a superb performer second in trackability only to the Type III. Finally, for optimum performance under a budget austerity program, the yeoman Shure M44E is for you. All in all, these are three great ways to enjoy music with the kind of system you have decided is best for you.

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SHURE ®

Trackability --1973

J. Kogen, B. Jakobs, F. Karlov*

In 1966, we wrote an article for Audio entitled "Trackability." That article described the concept of trackability as the ability of the stylus to stay in contact with the record groove at all times during the playing of the record. It was emphasized that the necessity of maintaining contact between the stylus and the record is basic to phonograph reproduction and must be considered as one of the major factors in the evaluation of a phonograph cartridge. In the ensuing years, this concept has been accepted throughout the world, and measurements of trackability are made in one form or another by almost all testing laboratories.

During the years since that presentation, a continuous research program on trackability and allied subjects has been under way in the Shure laboratories. The work has been centered about three primary areas of activity. First, a means of accurately measuring the trackability of phonograph cartridges has been developed. This was described by Anderson and Jenrick in Audio (August, 1972), and a trackability test record, TTR103, has been made available. Second, studies have been made on commercially available phonograph records to determine the trackability requirements that are imposed on the playback system. And, finally, a significant effort has been expended toward finding ways of improving the trackability of phonograph cartridges.

In this article, we will review the second and third areas of our investigative program. We will describe the results of measurements made on typical phonograph records. This, in essence, defines the tracking requirement imposed by the records, which must be accommodated by the playback system. We will then describe a design approach that satisfies these requirements.

Earlier Considerations of Trackability

The accepted definition of trackability has been: "The ability of the stylus to maintain contact with the record groove at minimum tracking force and at all frequencies within the audible spectrum."

*Shure Brothers, Inc.

This definition, which has been in use for many years, contains the term "audible spectrum." Our studies indicate that consideration must be given to frequencies outside the audible spectrum, and we will, therefore, modify the definition to: "The ability of the stylus to maintain contact with the record groove at minimum tracking force and at all frequencies found on phonograph records." The reasons for modifying the definition to include frequencies outside the audible spectrum will be discussed in detail later in this article.

In the 1966 article, reference was made to the limits of recorded modulation on records. In that reference, the audible frequency spectrum was broken down into parts and modulation limits applied as follows:

A. Low frequency, extending from 20 to about 800 Hz, with a maximum amplitude of groove modulation of 0.005 centimeters (0.002 inches).

B. Mid frequency, extending from about 800 to 2500 Hz, with a maximum modulation velocity of 25 centimeters per second.

C. High frequency, above 2500 Hz, with an acceleration requirement in excess of 400 g.

These limits were determined by some theoretical considerations as well as practical limitations in cutting phonograph records.

One of the practical limitations relates to the objective of providing a maximum playing time on each side of a record. This objective imposes a limitation on the amplitude of groove modulation. If the groove modulation is too large, it will not be possible to fit enough grooves onto the record to achieve this goal. In examining hundreds of records, we found that very rarely is the practical limit of 0.005 centimeters (0.002 inches) exceeded.

Another practical restriction is determined by the shape of the cutting stylus. The stylus is wedge-shaped (Fig. 1), and the relief angle between the sides and the front of the wedge determines the maximum angle the stylus is capable of cutting. This, in turn, determines the maximum modulation velocity to be found in the record groove. Near the inner grooves, this

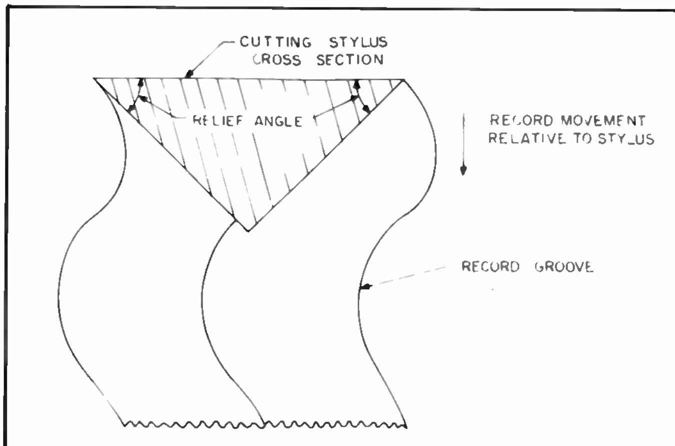


Fig. 1—Top view of cutting stylus in record groove.

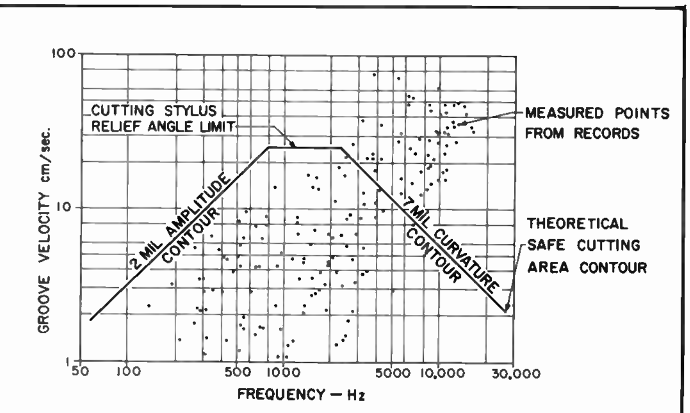


Fig. 2—Scatter diagram of "difficult-to-track passages" from commercial records.

limit is about 25 centimeters per second. While higher velocities can be achieved with specially designed cutting styli and at larger distances from the center of the record, we have rarely found modulation velocities greater than 25 centimeters per second in the 800 to 2500 Hz frequency range.

A theoretical limitation in the high-frequency region is based on the assumption that records should not be cut with a radius of curvature of modulation less than 0.7 mils (0.0007 inches). The reasoning for this is that such modulation would cause excessive distortion when the record is played with a stylus having a tip radius of 0.7 mils. In 1966, we stated that many records had been found to exceed this theoretical limit. In the next section, we will review our most recent findings on this subject.

Tracking Requirements Imposed by Commercially Available Records in the Audible Frequency Range

A study was made of a large collection of "difficult-to-track" records in order to update our information on trackability requirements in the audible frequency range. The results of this study are shown in Fig. 2.

Many of the records included in this study were suggested to us by critics and customers, while others were discovered by our engineers. These records contain some of the most difficult-to-track passages found in phonograph records. The records are all of good quality, worth having in one's collection, but with characteristics that for one reason or another make them difficult to play. It was felt that this collection would constitute a reasonable sampling of problem records that a top-quality phonograph system should be able to reproduce with minimum distortion.

On each of these records, difficult-to-track passages were identified, and the modulation velocity and frequency for each passage were measured. Each of these measurements was then plotted as a point in Fig. 2. The envelope of the points presents a picture of the maximum recorded velocities found in this sampling of phonograph records illustrated as a function of frequency.

One point of information that should be interjected here is that some of the points in Fig. 2 would not appear to be difficult to track in terms of modulation velocity and frequency. The measurements we have made assume single-frequency signals. Many of these passages actually contained a number of signals at different frequencies, the dominant one being present in Fig. 2. The total trackability requirement is actually the sum of those imposed by all of the frequencies acting simultaneously. It is also probable that the existence of warps in some of these records aggravated the tracking problem.

Superimposed on the scatter diagram of points in Fig. 2 are the "quasi" theoretical maximum velocity limits one would expect for recorded modulation velocity. Note that in the low-frequency region, from 20 to 800 Hz, and in the mid-frequency region, from 800 to 2500 Hz, none of the measured points exceeds this curve. This is in agreement with the statements made previously regarding the 0.005-centimeter (0.002-inch) amplitude limit and the 25 centimeters-per-second velocity limit in the low- and mid-frequency regions. The theoretical high-frequency limit is exceeded by many points, however, and it seems reasonable to assume that records have been and will continue to be cut with very high modulation velocities in the high-frequency region.

There are several reasons for the existence of high-velocity program material in the high-frequency region of phonograph records. High-frequency, high-level signals provide a means of broadening the dynamic range and producing impressive high-frequency sounds from records. Also, no significant mechanical or electrical restriction is placed on the cutting equipment in this region. It is obvious from the points on Fig. 2 that cutters are capable of achieving very high modulation acceleration. The only major restriction that the recording engineer apparently considers is the ability of the cutter to inscribe this high-level, high-frequency program material without "burning out" the equipment!

The conclusion reached from the reexamination of trackability requirements in the audible frequency region is that the major problem area still lies in the high-frequency region. The theoretical limits in the low- and mid-frequency regions are reasonably well defined and held. In the high-frequency region, at this point in time, our objective appears to be one of maximizing trackability with no clear limit as to the maximum that must be achieved.

Evaluation of Commercially-Available Records—Subaudible Frequencies

The tracking requirements imposed by the record can be significantly affected by the existence of warps and thickness variations in the record. Consider, for example, one of the difficult-to-track records we measured. The trackability requirement of the modulation in the record groove may be sufficient to drive the phonograph cartridge to the limit of its ability to play that modulation on a perfectly flat record. In such a case, all the available stylus force is used up just to play the groove modulation. Now, think of what will happen if the record is warped. In addition to being required to play the high-modulation velocity in the record, the cartridge must also follow the warp undulation. This will cause additional stylus force to be used up. The addition of the warp, therefore, could be enough to cause mistracking because of a lack of available stylus force. Since high modulation velocity and warp can and do happen simultaneously, it is very important for us to learn more about warp of commercially-available records.

Warp appears on records primarily in the form of subaudible frequencies—that is, below 20 Hz. A study was performed to determine the amplitude as well as the frequency band in which warps occur. The method involved was to examine a significant number of randomly-chosen records by making measurements of the surface contour variations. These contour measurements were than analyzed to determine the amplitude and approximate frequencies of the warps.

A measuring technique was developed that provided a profile of the record surface as a function of angular position around the circumference of the record. Figure 3 presents a series of typical profiles. The vertical axis shows amplitude of the warps in thousandths of an inch; the horizontal axis shows angular position in degrees for 360 degrees around the record i.e., a full revolution. Figure 3 also shows several examples of

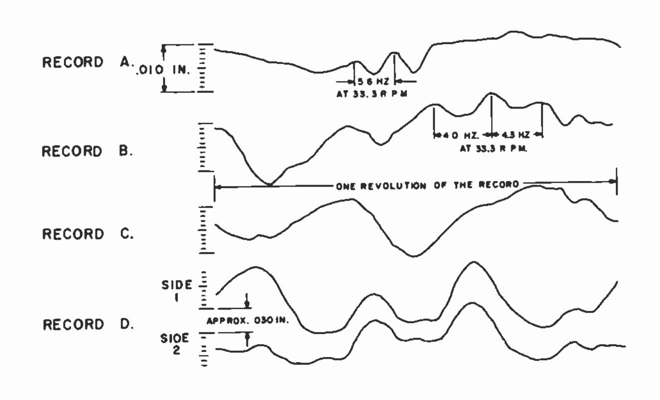


Fig. 3—Examples of actual surface contours found on commercial phonograph records.

frequency approximations, such as the section in record "A" around the 180-degree point, where an amplitude of about 0.005 inches and a frequency of 5.6 Hz are indicated.

Some of the records were measured on both sides, so that with proper alignment, thickness variation could also be determined. An example of this is shown in part "d" of Figure 3, in which the distance between the two curves indicates record thickness. In this particular record, there is a fairly significant variation in thickness. While it was found that thickness variations can be significant, their effect on disturbing the record surface profile is indistinguishable from warps as far as the playback problem is concerned. We will not, therefore, consider thickness variations as phenomena separate from warps.

In the study, 67 randomly-selected records were measured and 210 warp conditions were identified. Figure 4 is a scatter diagram that shows each of these points plotted at the appropriate amplitude and frequency. The dashed line across the bottom of the diagram represents the maximum recorded groove modulation amplitude limit—that is, 0.005 centimeters (0.002 inches). The solid curve lying above all the points represents the maximum expected warp amplitude based upon this random selection of records. The solid curve will be assumed to represent the worst warp conditions the playback system will be required to cope with in the range below 20 Hz.

The figure indicates that the warp amplitude increases as the frequency decreases. While we will not attempt to discuss this in detail here, another consideration is that of the probability that such warps will occur. In our study, we found that a large number of serious warps occur at frequencies around 4 Hz. This would indicate that although the maximum amplitude of warp occurs at the lowest frequency—that is, one-half Hz—the problem is really most significant in the region of approximately three to five Hz, where one must combine the problem posed by the amplitude of the warp along with the probability that such a warp will occur more frequently.

We should also state that the 67 records randomly chosen for this study were pressed within the past few years and include samples from most of the large record companies throughout the world. They should, therefore, be representative of the typical audiophile's record collection. The maximum warp curve of Fig. 4 could be subject to modification in the future if means should be developed for improving the flatness of records. We feel that for the present, however, this study gives us a reasonable, practical perspective of record warps and the challenge they present to the playback systems.

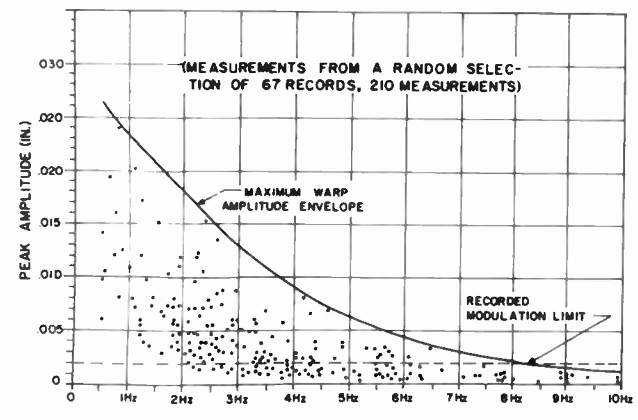


Fig. 4—Scatter diagram of warp amplitude and frequency found on commercial records.

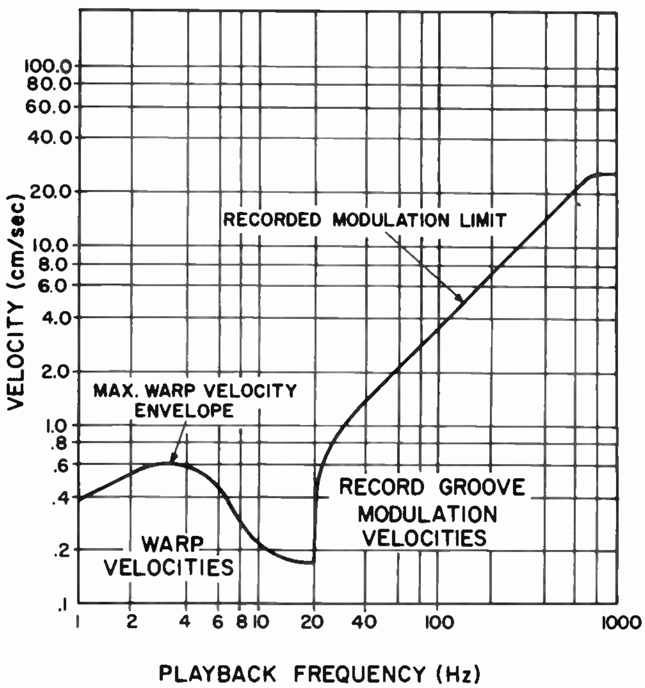


Fig. 5—Extended low frequency trackability requirements.

Trackability Requirements in the 0.5- to 20,000-Hz Range

The data shown in Fig. 4 can be used to present an extended low-frequency trackability requirement for phonograph cartridges. Figure 5 shows the trackability requirements extended into the subaudible region to encompass the warp frequencies. The objective in the design of a phonograph cartridge must be to provide trackability that exceeds the limits shown in Fig. 5. Several factors should be considered in attempting to meet this objective and also to provide the required high-frequency trackability.

In the very low-frequency region below approximately 100 Hz, trackability is determined by the tone arm and the phonograph cartridge operating as a system. In this region, trackability is not controlled solely by the cartridge as it is in the audible spectrum above approximately 100 Hz. We cannot specify trackability for the phonograph cartridge alone but must consider the cartridge in combination with the tone arm.

Above approximately 100 Hz, trackability is determined solely by the phonograph cartridge. In phonograph cartridge designs, it is possible to "trade off" trackability in the low-frequency region in order to increase trackability in the high-frequency region. It is possible to provide only enough low-frequency trackability to satisfy the low-frequency requirements and to place the major emphasis in the high-frequency region. The design is wasteful, to say the least, if excessive low-frequency trackability margin is provided with a resulting reduction in high-frequency trackability.

Since recorded modulation at all frequencies and warps can occur simultaneously, it is necessary that the cartridge-arm system be able to track properly throughout the total significant frequency band. It is not sufficient for the cartridge-arm system to be capable of tracking audio frequencies and yet be incapable of properly coping with the warp modulation. The objective in the design of the phonograph cartridge must be to resolve both of these problems at the same time.

Optimizing for Low-Frequency Trackability

In the low-frequency region, the amplitude of the record groove excursion caused both by the recorded groove modulation and by the warp of the record is the major factor to be considered. In order to track in the low-frequency region, it is necessary for the stylus to be capable of deflecting to the maximum amplitude with the available tracking force. At low frequencies, this ability is determined primarily by a parameter that we call the dynamic compliance of the stylus. The compliance figure normally specified for phono cartridges in "10^{-6} centimeters per dyne" is the static compliance; that is, the compliance that would be measured under static conditions. Dynamic compliance is measured while the stylus is in motion and is normally smaller than the static compliance.

At some low frequency, the tone arm-cartridge system will exhibit a resonance determined primarily by the interaction of the dynamic compliance of the stylus and the effective mass of the tone arm-phono cartridge combination. At and near the resonance frequency, the motion of the stylus relative to the tone arm will be many times that of the exciting signal. This will then significantly reduce the ability of the pickup to track the groove modulation. It is essential, therefore, that we minimize as far as is practical the probability of exciting this resonance.

To further examine the effect of tone arm-cartridge resonance, let us consider an experiment as depicted in Fig. 6. Here we have a tone arm and cartridge mounted with a device that can drive the stylus at a constant amplitude with varying frequency. Such a device could be the coil-magnet assembly of a loudspeaker. We will measure the output of the phonograph cartridge as the frequency is varied from a very low frequency up to 100 Hz. We will assume for the purpose of this example that this particular system resonates at 10 Hz.

At one-half Hz, the tone arm and cartridge move up and down together as a unit. There is little relative motion between the stylus and the tone arm. Since the stylus moves only slightly with respect to the tone arm, little electrical signal is generated and we can plot point 1 of Fig. 7.

As the frequency of the input signal is raised, we begin to discern more relative motion between the stylus and the tone arm. Keep in mind at this time that the total system is moving up and down with the driving device, and we are now obtaining an additional motion of the stylus relative to the tone arm. This produces some output, as shown at point 2 of Fig. 7.

As the system approaches resonance, the tone arm moves in ever-increasing amplitude, reaching a violent motion at 10 Hz. There is considerable relative motion between the stylus and the tone arm, and considerable electrical output from the cartridge (point 3, Fig. 7). This is the resonance frequency.

As the frequency is raised above 10 Hz, the relative amplitude of motion between stylus and tone arm decreases, and at about 40 Hz becomes constant with increasing frequency. At this point, the tone arm is standing still and all of the motion is taking place in the stylus. This is the condition under which the phonograph system is supposed to operate in the recorded frequency range. At these frequencies, one cannot see movement of the stylus with the naked eye.

There are two very important observations to be made from this experiment. First, the movement of the cartridge-tone arm system at and around resonance can cause considerable difficulty in tracking, and the frequency at which this resonance occurs must be determined with great care. Second, motion of the cartridge-tone arm below, say, 20 Hz is an indication that the system is having some difficulty. It is important to minimize the possibility and probability of generating such motion.

Optimizing the Resonance Frequency of the Cartridge-Tone Arm System

The optimum resonance frequency for the cartridge-tone arm system can be deduced from Fig. 5. The figure shows that the mimimum amplitude for warp and recorded groove modulation is in the region around 10 Hz. By measuring the cartridge-tone arm system at low frequencies in a manner similar to that just described, we can determine the low-frequency trackability of the system. A typical low-frequency cartridge-tone arm trackability curve is shown in Fig. 8, which also repeats the warp and recorded groove modulation velocity requirements of Fig. 5. In order to optimize the system, we should place the point of minimum trackability (the resonance frequency of the tone arm-cartridge system) in the region of minimum warp and record modulation velocity; that is, around 10 Hz.

The two major factors that affect the resonance frequency are the mass of the tone arm-cartridge system and the dynamic compliance of the stylus. Increasing the mass of the system will lower the resonance frequency and tend to raise the amplitude of the resonance, or, conversely, decrease the trackability. Increasing the dynamic compliance will also decrease the resonance frequency. By adjusting these two parameters, one can find an optimum frequency for the minimum point of the trackability curve. This optimum frequency is in the range between 7 and 15 Hz where there is a minimum of groove modulation and warp input to excite system resonance.

From the standpoint of phonograph cartridge design, it is necessary that the cartridge be made to operate with available good quality tone arms. A survey of tone arms indicates that the effective mass can be expected to fall in the range of 13 to 30 grams. (This includes typical cartridge mass.) This, then, sets a definite restriction on the dynamic compliance of the cartridge. Using the information obtained as to the optimum resonance frequency and the range of effective tone arm masses, we can calculate the optimum dynamic compliance for a phonograph cartridge to be in the range of 20 to 25 microcentimeters per dyne (10^{-6} cm/dyne).

The conclusion we have reached is most significant with regard to optimizing the design of a phonograph cartridge. We

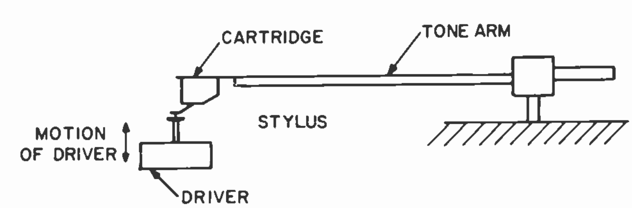


Fig. 6—Method for measuring tone arm-cartridge resonance.

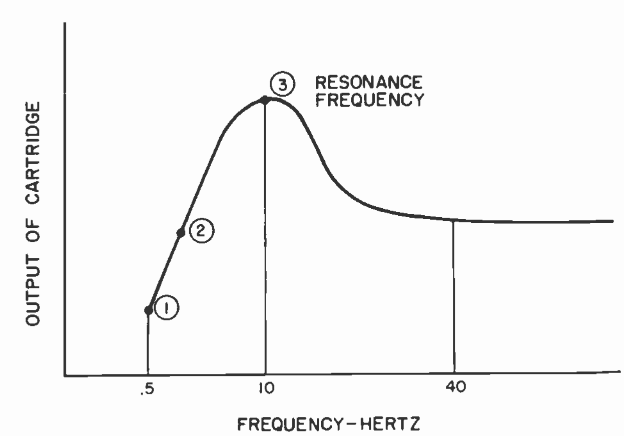


Fig. 7—Output of cartridge as drive frequency is changed near tone arm-cartridge resonance.

have found that there is actually an optimum compliance, and that either too much or too little compliance can lead to difficulty.

If the compliance is too high, the resonance frequency of the tone arm-cartridge system will be too low; and severe problems will result when the system is used with warped records. Aggravated wow and groove jumping are typical in this case. In addition, large amplitude, subaudible electrical signals will be generated. The signals can easily overload an amplifier or cause excessive stress and possible distortion in the loudspeaker.

If the stylus compliance is too low, causing the resonance frequency to be too high, several other problems can occur. Very low audio frequencies may be over-emphasized because of the rise in response near resonance. One might also expect increased mechanical and acoustical feedback problems. And, finally, problems may occur because of insufficient low-frequency trackability in the audio region.

The seriousness of the low-frequency resonance problem should not be underestimated. We have found that the sensitivity to record warp resulting in mistracking and wow can differ by as much as a factor of ten between properly and improperly designed tone arm-cartridge systems.

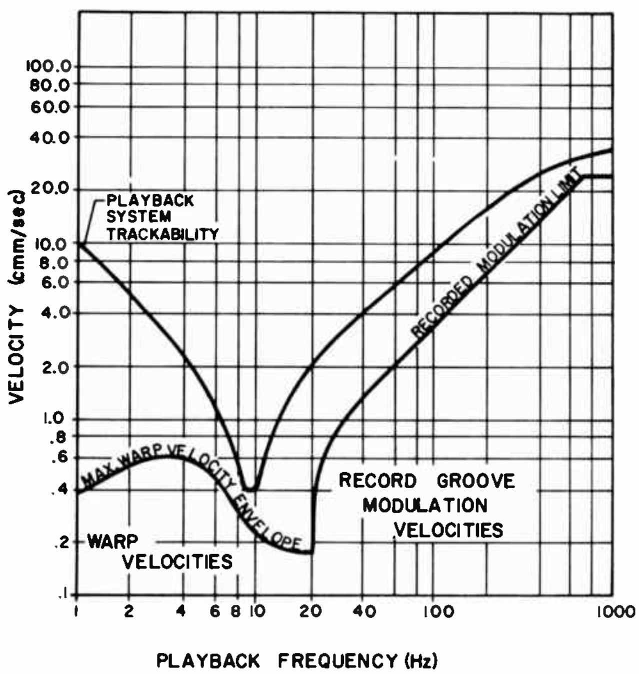


Fig. 8—Typical tone arm-cartridge trackability curve.

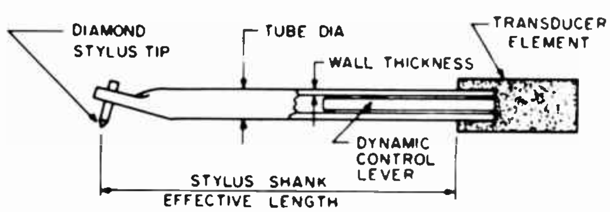


Fig. 9—Stylus of Shure phonograph cartridge.

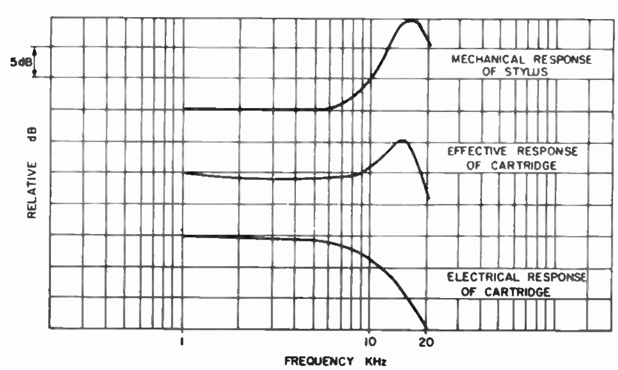


Fig. 10—Mechanical, electrical, and effective frequency characteristics of V15-I.

Optimizing for High-Frequency Trackability

The need for maximizing high-frequency trackability has been established (Fig. 2). Our objective in the high-frequency region is to obtain all the trackability possible, provided that we have allowed sufficient tracking capability in the subaudible, low-, and mid-frequency regions. The studies that yielded the data illustrated in Fig. 2 give information as to the required trackability for the cartridge. Other studies relating to evaluating cartridges, as reported by Anderson and Jenrick in Audio (August, 1972), resulted in the TTR103 Test Record, which is now commercially available. This record allows measurement of trackability in the low-, mid-, and high-frequency regions. The remaining and most critical objective is to determine how to design a cartridge that will satisfy the known trackability requirements.

The major factor in maximizing high-frequency trackability of the cartridge is the design of the stylus. When studied in detail, the stylus is a complex mechanical structure. In previous papers, this subject was discussed in some detail, and we will not repeat that information here. Suffice it to say that development and design engineers must pursue a task of considerable complexity, using both analog and digital computational techniques, along with arduous and painstaking development and evaluation of prototypes. Parameters that must be optimized include many stylus dimensions; stylus shank, wall size and thickness; tip shape and size; magnet geometry; different materials; and—in the case of the Shure cartridge—dimensions and positioning of the dynamic control lever (Fig. 9). The general direction in which the engineer moves is that of decreasing the effective stylus mass. However, there are many limitations on this objective, not the least of which is ruggedness and reliability. Output level and clearance between the cartridge and the record surface are other factors that must be considered when reducing the stylus mass.

While one primary objective is to maximize high-frequency trackability, we must not forget other very important characteristics of the phonograph cartridge. It is possible to improve trackability, but at the same time deteriorate the frequency response. There is a point beyond which this is not acceptable. A brief historical review will be useful to explain this point.

The frequency response of a phonograph cartridge results from the combination of two frequency characteristics: the mechanical characteristic of the stylus and the electrical characteristic of the cartridge. Figure 10 shows these characteristics for the original Shure V-15 I phonograph cartridge, which was introduced in 1964. In the figure, we show the effective frequency response, as well as the relative responses of both the stylus and the cartridge alone. Addition of the mechanical and electrical curves results in the effective response. Since trackability is generally poor at resonance, it is clear that the V-15 I would have difficulty tracking material in the 15 kHz region.

Trackability of the Shure V-15 II cartridge was substantially improved by moving the mechanical resonance frequency to 20 kHz while many other parameters of the cartridge were idealized. In the design of the V-15 II, the decision was made to accept a small droop in the frequency response in exchange for the vastly improved trackability. This is shown in Fig. 11.

In the development of the newly introduced Shure V-15 III (Fig. 13), one primary objective was to further improve the high-frequency trackability. In the process of making this improvement, the stylus resonance was moved out to 23,000 Hz, well above the audible spectrum. It was clear, however, that with this additional raising of the resonance frequency, a considerable droop in the frequency response curve would occur if the cartridge body of the V-15 II were retained. It was thus necessary to design a completely new cartridge body structure to match the higher trackability stylus of the V-15 III.

The cartridge structure of the V-15 III incorporates an assembly of precise miniature laminations. These laminations reduce electrical losses in a manner similar to the laminations of an electrical transformer. Through the use of these laminations and several other design features, it is possible to provide an electrical frequency response that almost perfectly complements the mechanical frequency response of the stylus. This results in an overall frequency response that is essentially flat (Fig. 12). Through the use of this new structure, it is possible, therefore, to achieve a significant improvement in high-frequency trackability and, at the same time, to provide an improvement in frequency response over that of the previous design—the V-15 II.

The trackability curve of the V-15 III as compared to the V-15 II is shown in Fig. 14.

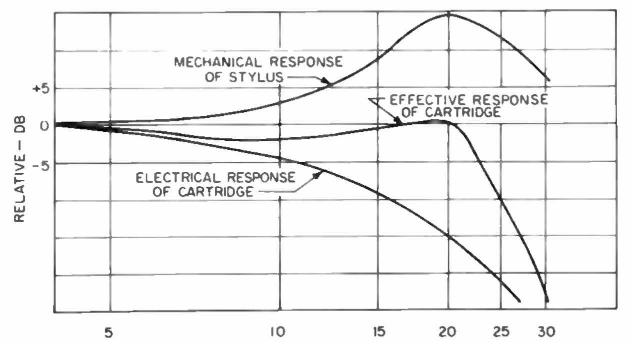


Fig. 11—V15-II frequency response characteristics.

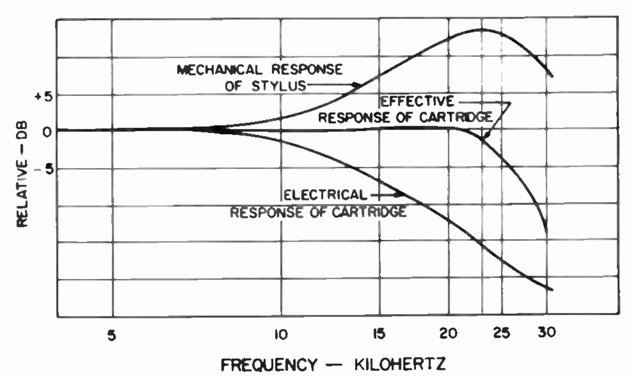


Fig. 12—V15-III frequency response characteristics.

Conclusion

New product development must always be a matter of continuous progress. In many instances, however, we must retrace our steps, reexamine our previous results, and provide improvements before moving ahead. While such factors as record warp and thickness variations, and high-velocity groove modulation have been known to us for many years, we have found it valuable to learn more about them. We have learned about them by developing measuring techniques and by performing extensive studies, both on discs and on cartridges.

Our measurements show that warp is a significant problem in current phonograph records and that it is essential to optimize the dynamic compliance of a phonograph cartridge so that records may be played properly. Since warp exists on practically all phonograph records to one degree or another, it is an important characteristic that cannot be ignored. Fortunately, we know how to minimize the effects of warp without compromising the low-frequency response. Only severely warped records need cause significant problems.

We have also learned that through careful optimization, it is possible to design a phonograph cartridge that satisfies the high-frequency tracking requirements of practically all phonograph records and, at the same time, offers an almost perfect, flat frequency response. The art and technology of making records has progressed significantly since the report we presented in AUDIO in 1966. We believe that the design of phonograph cartridges and their ability to play the records has kept pace with that progress. Æ

Fig. 13—V15-III

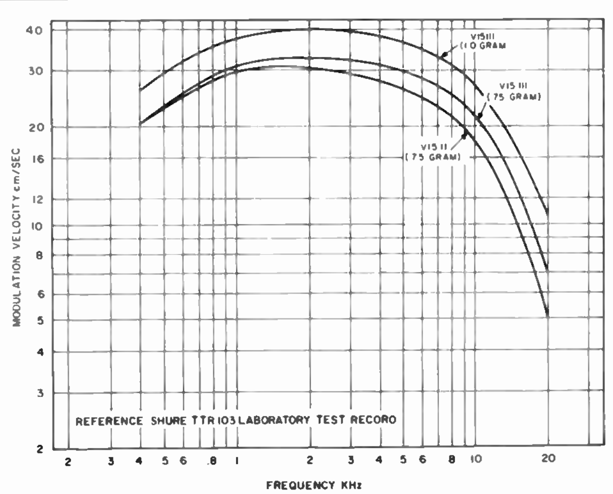
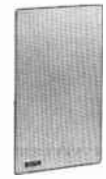


Fig. 14—Trackability of V15-II and V15-III.

Acoustic Research AR-8

Akai SW-161



New Advent

Speakers

MANUFACTURER	MODEL	WOOFER: Diameter, in.	WOOFER: Resonance (in system), Hz	WOOFER: Enclosure type	MID-RANGE: Diameter, in.	MID-RANGE: Type	TWEETER: Diameter, in.	TWEETER: Type	Overall freq. resp., Hz to kHz ±? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont), W	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
ACOUSTIC RESEARCH	LST*	12	42	Acous. susp.	(4) 1½	Hem; dome	(4) ¾	Hem; dome	**	50	***	575; 5000	4	27½ x 20 x 9¾	Wal.	Cloth	90	600.00 ea.	*6-pos. swit ; contld. freq. resp. crvs.; ** compl data avail. on req.; *** data avail. on req., var.
	3a	12	42	Acous. susp.	1½	Hem; dome	¾	Hem; dome	**	25	***	575; 5000	4	14 x 25 x 11½	Wal. unf	Cloth	53	269.00 ea.	**Compl data avail on req.; ***var., data avail. on req.
	5	10	56	Acous. susp.	1½	Hem; dome	¾	Hem; dome	**	20	***	650; 5000	8	13¼ x 24 x 11¼	Wal. unf	Cloth	39	189.00 ea.	Same as above.
	2ax	10	56	Acous. susp.	3½	Cone	¾	Hem; dome	**	20	***	1400; 5000	8	13½ x 24 x 11½	Wal. grd	Cloth	36½	139.00 ea.	Same as above
	8	10	54	Acous. susp.			1½	Cone	**	15	***	1800	8	13½ x 24 x 11½	Wal. grd	Cloth	32	119.00 ea.	Same as above.
	6	8	56	Acous. susp.			1¼	Cone	**	15	***	1500	8	12 x 19½ x 7½	Wal. grd	Cloth	20	89.00 ea.	Same as above.
	4xa	8	65	Acous. susp.			1¼	Cone	**	15	***	1600	8	10 x 19 x 9	Wal. grd	Cloth	18½	75.00 ea.	Same as above.
	7	8	70	Acous. suso.			1¼	Cone	**	15	***	2000	8	9¾ x 15¾ x 6¼	Wal. grd	Cloth	11	65.00 ea.	Same as above.
ACOUSTICAL MFG.	Quad ES			Doublet ES					45-18k ± 5	28				34½ x 10½ x 31		Alum. blk.	40	345.00 ea.	Full range ES.
ADVENT	Advent		43	Acous. susp.			⅞	Dome	30-20k ± 4	20	*	1k	8	14¼ x 11½ x 25½	Wal.	Cloth, light	44	120.00	Also in wal. vinyl, $105.00; *avail. upon request from Advent.
	Smaller Advent		43	Acous. susp.			⅞	Dome	30-20k ± 4	15	*	1.4k	4	11½ x 9¼ x 20	Wal. vinyl	Cloth, light	26	72.00	*As above.
	New Advent		57	Acous. susp.				Cone	40-20k	10	*	1.5k	8	11½ x 7¼ x 19	White	Metal, nat.	17	58.00	*As above.
AKAI	SW-175	15	20		5¼		2¼; 3; 3½	Horn Cone Dome	20-23k	4-6	25-30	600; 5K; 10K; 15K	8	17 x 24.4 x 11.2	Wal.	Lattice work	49	295.00	
	SW-161	12	30		6½		2 1	Horn Dome	30-20k	4-6	25-30	600; 5K; 15K	8	16.2 x 12 x 25.6	Wal.	Wal. grille	42.9	200.00	Ind. hi & midrng. lev. contls.; mag. cir.; 6½" non-leak midrng.
	SW-155	12	30		5	Cone	2½ 2	Dome Horn	25-21k	4-6	25-30	1000; 5K; 15K	8	16 x 11¾ x 25	Wal.	Wal. grille	38.9	175.00	Hi & midrng. contls.; extra wide freq. resp.
	SW-135	10	25		5	Cone	2½	Alum. voice coil	40-20k	4-6	25-30	1200; 6k	8	13.2 x 10.8 x 21.2	Wal.	Wal. grille	18	130.00	Tone sel.; spkr. jack & push but. term.
	SW-125	10	30		5	Horn	2	Horn	40-20k	4-6	25-30	1200; 5000	8	13⅞ x 21 x 11⅝	Oil wood	Oil. fin. grille	24.2	145.00	
ALTEC	887A	8		Acous. susp.			3	Cone		12	45	2.5k	8	10 x 9 x 19	Wal.	Cloth, dark	17.8	75.00	
	891A	12		Acous. susp.			2½	Cone		12	60	1.6k	8	14½ x 12⅛ x 25½	Wal.	Foam, blk.	35	129.00	
	Concept EQ	12		Acous. susp.	5	Cone	2	Dome		30	100	800; 4k	8	15 x 10⅝ x 25	Wal.	Foam, blk.	38	180.00	
	878B	15		Acous. susp.			*	*		30	60	800	8	26⅝ x 18½ x 30⅛	Wal.	Foam, blk.	110	450.00	*Voice of the Theatre comp. horn driver.
AUDIOANALYST	A76	10	52	Acous. susp.			1¾	Cone	44-18k ± 3	15	50	2500	8	12¼ x 11⅛ x 21	Wal. Vinyl	Foam	32	79.00	Linear resp.
	A100	10	48	Acous. susp.	3	Cone	2	Cone	40-20k ± 3	15	60	1500; 7500	8	13¾ x 12 x 24⅝	Wal.	Cloth brn. & blk.	37	129.00	Linear resp.
	A200	12	46	Acous. susp.	5	Cone	(1) 3½; (2) 2	Cone	38-20k ± 3	15	100	500; 2000; 7500	8	15 x 12¾ x 27	Wal.	Cloth blk.	53	225.00	Angled supertweeters.
	Pyra-media	10			(2) 5	Cone	(6) 1¾	Cone	36-20k ± 3	20	100	600; 2500	8	30 x 30 x 46			80	575.00	Omnidir./adj. disp.; incl. functional pedestal.
AUDIO DYNAMICS	450A	12		Acous. susp.				Dome	25-30k ± 3	10	65		8	14 x 12⅝ x 25	Wal.	Cloth; char.	50	150.00	
	303ax	10		Acous. susp.			2	Cone	37-20k ± 3	10	45	1.5k	8	13 x 11¾ x 23¾	Wal.	Cloth; blk.	37	100.00	Mid & hi contls.
	303b	10		Acous. susp.					37-20k ± 3	10	45	1.5k	8	12 x 11 x 21¾	Wal.	Cloth	37	85.00	

Be Sure To Hear The Advents.

If you are thinking of buying some stereo equipment, Advent products are worth looking for and listening to.

There are two Advent loudspeakers (the original Advent Loudspeaker and The Smaller Advent Loudspeaker) and two Advent cassette decks (the Models 201 and 202). They are

all best sellers in their respective categories, although they have been advertised relatively little and are sold only through a limited number of dealers, chosen for their ability to understand and display what they are selling.

The reason for their popularity is that they do what they are represented to do. Pleased customers go out of their way to tell us and others about that.

Both Advent Loudspeakers are designed to provide the very top level of speaker performance at a fraction (half or less) of the former going cost. The only difference in performance between the two is that the original will play slightly louder in bigger rooms than The Smaller. Either of them make it possible to put together as satisfying a stereo system as you are ever likely to want for a reasonable, thoroughly affordable amount of money. The original costs $105-$125, depending on cabinet finish and the part of the country it's shipped to; The Smaller costs $70-$75.

Both of them are intended for direct critical comparison with the most expensive and elaborate speakers available.

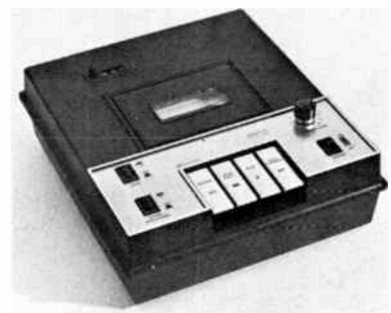
The Model 201 cassette deck has been called *the* best and *the* state-of-the-art cassette machine by audio reviewers, who are seldom that explicit about a product. It is the most satisfying tape machine of any kind that most music listeners can buy. It can make and play cassette recordings that equal or surpass the sound quality of the best records, and its performance also compares easily with that of far more expensive and complex open-reel tape recorders. Since cassettes are far

The 201

easier to handle than records, and subject to far less wear and tear in normal use, the 201 can add an enormous amount to the pleasure of listening to music at home. It costs $280.

The 202 is a deck designed solely to play back cassettes. It is for people who would like the equivalent of a turntable for commercially recorded cassettes, either because they don't want to record their own or because they already have a cassette machine that records. It costs $130.

We also are just about to market the first real innovation



The 202

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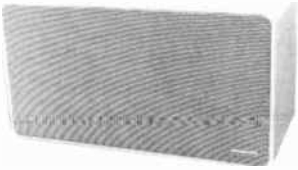
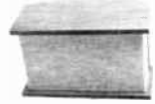
Speakers

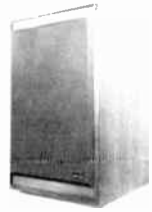
Audioson-Kirksaeter Monitor



Bose 901

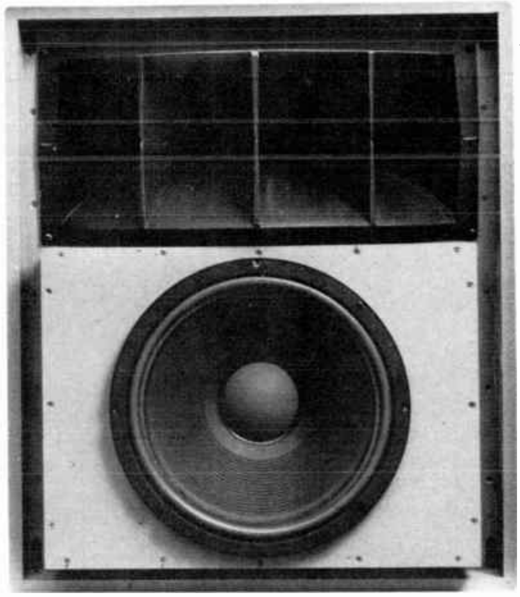


Cerwin-Vega 212



MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp., Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
AUDIO PROJECT	V			Acous. *	(9) 5 **	Cone		PZ x-tal	25-22k ±3	30	180	***	8	20 x 14 x 13	Wal.	Cloth, dk. brn.	43	500.00 pr	W. eqlzr.; *radiating; **full range; ***capacitorless.
AUDIO RESEARCH	Tympani I A								50-15k ±2	50-500	100	1-3k; 3200 built in	8	4'1" x 6' each	no	Cloth, sev. opt.	160/ pr.	1095.00 pr.	Magneplanar™ ldspkr.
AUDIOSON-KIRKSAETER	35	8		Acous. susp.	1½	Dome	1	Dome	30-20k	15	35	3500	4-8	18 x 8 x 9	Wal	Cloth, brn.	14	110.00	
	50	10		Acous. susp.	1½	Dome	1	Dome	22-22k	25	50	1200; 5500	4-8	18 x 9¾ x 11¼	Wal.	Cloth, brn.	22	180.00	
	70	12		Acous susp.	1½	Dome	(2) 1	Dome	20-22k	30	70	1200; 5500	4-8	22½ x 11 x 13	Wal	Cloth, brn	40½	260.00	
	100	(2) 10		Acous. susp.	(2) 1½	Dome	(2) 1	Dome	18-22k	35-50	100	1200; 5500	4-8	29½ x 12 x 16½	Wal.	Cloth, brn.	66	450.00	
AVIO	100	8	60	Acous.			1¾	Cone	40-18k ±5	10	75	2500		13 x 8⅝ x 24⅞	Wal	Knit fabric	22	79.50*	Fuse prot.; 3-pos. frnt mntd. bal. swit. *$5.50 higher in the West.
	102	10	47	Acous.			1	Dome	35-18k ±5	15	100	2200	8	15 x 9⅝ x 25	Wal.	Knit fabric	36	109.50*	*Same as above.
	103	10	47	Acous.	4½	Cone	1	Dome	35-18k ±5	20	150	500; 3500	8	15 x 9⅝ x 25	Wal.	Knit fabric	38	139.50*	*Same as above; 5-pos. frnt. mntd. high freq. bal. swit.; base incl.
B & O	5700	(2) 10		*	2½	Dome	¾	Dome	35-20k ±4	20	60	550; 7000	4-8	26 x 14¼ x 11¾	Rose	Cloth, blk	48	360.00 ea.	*Passive bass rad.
	4702	2 x 7½		Acous. susp.	5	Cone	1½	Dome	50-15k ±4	15	60	500; 3000	4	23 x 11½ x 11½	Rose	Cloth, blk.	29	220.00 ea	
	3702	8		Acous. susp.	3⅜	Cone	1	Dome	50-16k ±4	15	40	1000; 4000	4	19¾ x 10 x 10	Rose	Cloth, blk.	20	145.00 ea.	
	2702	7		Acous. susp.			1½	Dome	58-15k ±4	15	25	3000	4	16 x 8 x 8	Rose	Cloth, blk.	12	100.00 ea.	
B&W (Linear Devices)	70CA	13		Acous. susp.			(11)	ES	22-18k ±2	60	50	400	8	27 x 15½ x 32½	Wal./wte.	Cloth; brn. blk.	80	660.00	
	DM2a	8		Trans. line	1½	Dome	1	Dome	50-20k ±2	30	50	3.5k; 14k	8	14 x 13¾ x 25½	Wal./wte.	Cloth; brn. blk.	47	300.00	Opt. pedestal.
	DM4	8		Bass reflex	1½	Dome	1	Dome	50-20k ±4	15	25	3.5k; 14k	8	10 x 10 x 21	Wal./wte.	Cloth; brn. blk.	24	180.00	
BOSE	901			*	(9) 4½	Cone				25	270		8	20½ x 12⅞ x 12¾	Wal.	Cloth, bge. or brn.	33	525.00 pr.	*Dir./reflecting; no x-overs; active equalizer
	501	10	55	Acous. susp.			(2) 3	Cone		20	100	1700	4	14½ x 14½ x 24	Wal. vinyl	Cloth brn.	38	139.00	Dir./reflecting; floor standing.
CERWIN-VEGA	Magnastat I	12	38	duct. refl.	4x4	*			30-25k ±2.5	2	100	250	4-8	15 x 15½ x 29	Wal.	Cloth, choice		398.00 ea.	*4x4 rad. dia
	212	12	40	duct. refl.	5	Cone	2½	Dhorn	35-20k ±3.5	1	100	300; 5000	4-8	15 x 15½ x 26	Wal	Cloth, choice	60	199.50 ea.	
	320 Mod. Sys.	15 or (2) 12	34	duct. refl.	12	Cone	(2) 2½ 2x6	Dhorm Horn	30-20k ±3	½	150	125; 1500; 4000	8	Mt- 18 x 6 x 16¾ B-20 x 25 x 25	Wal.	Cloth, red & blk.		499.00 ea.	
	26	12	43	duct. refl.			2½	Dhorm	30-25k ±4	2	60	2500	4-8	14½ x 12 x 25	Wal.	Cloth, choice	55	279.00	
CREATIVE	90	12		Acous. susp.	10	Horn	2½		30-20k		40	4000; 8000	8	14¾ x 14¼ x 23¾			41	159.95 ea.	Mid and hi contls.
	99	15		Acous. susp.	5		1	Dome	30-20k		55	700; 3000	8	20¼ x 17 x 25½	Wal.		70	225.00 ea.	Opt. marble or smkd. glass top. $249.95.
	100	12		Acous. susp.	5		1	Dome	30-20k		50	700; 3000	8	14¾ x 12¼ x 23¾			42½	189.95 ea	
	200	10		Acous. susp.	5		1	Dome	30-20k		40	700; 3000	8	12¾ x 12¼ x 22			27	99.95 ea.	
CRISMAN	Bookbinder Mk II	8	55	Acous. susp.			5x2	Horn	50-18k	10	35	3500	8	21 x 12 x 12	Wal.	Cloth, choice	35	98.00 ea	Cont. adj. x-over contl.
	Glendinning Mk V	12	40	Acous. susp.	10x4	Horn	5x2	Horn	40-18k	7	40	1000; 3500	8	29 x 15 x 15	Wal	Cloth, choice	65	165.00 ea	2 cont adj. x-over contls.
	Heffalump Mk VII	15	35	Acous. susp.	10x4	Horn	5x2	Horn	35-18k	8	50	1000; 3500	8	31 x 18 x 18	Wal.	Cloth, choice	83	198.00 ea.	
	Henry III Mk IX	15	25	Refl.	12 x 5	Horn	8x3	Horn	25-23k	15	90	800; 3500	8	31 x 21 x 18	Wal	Cloth, choice	120	498.00 ea.	

MONITOR



This is an official Altec studio monitor loudspeaker—the 9846-8A. It's called a monitor because it's designed for just one job: to deliver the purest, most accurate possible definition of every detail of every sound. In a recording studio, definition of detail is a must. Detail that differentiates instruments from the very lowest to the very highest frequencies. Detail that differentiates various models of microphones—for each has its own sound pick-up characteristic. Detail that differentiates microphone/instrument distances. In the close-miked world of contemporary music, a foot either way can make a lot of difference.

Low distortion in a studio monitor is also a necessity. It prevents fatigue that sets in after long periods of high volume listening. And short bursts of sound must be captured instantaneously ("transient response") to avoid mushy reproduction that results in loss of detail.

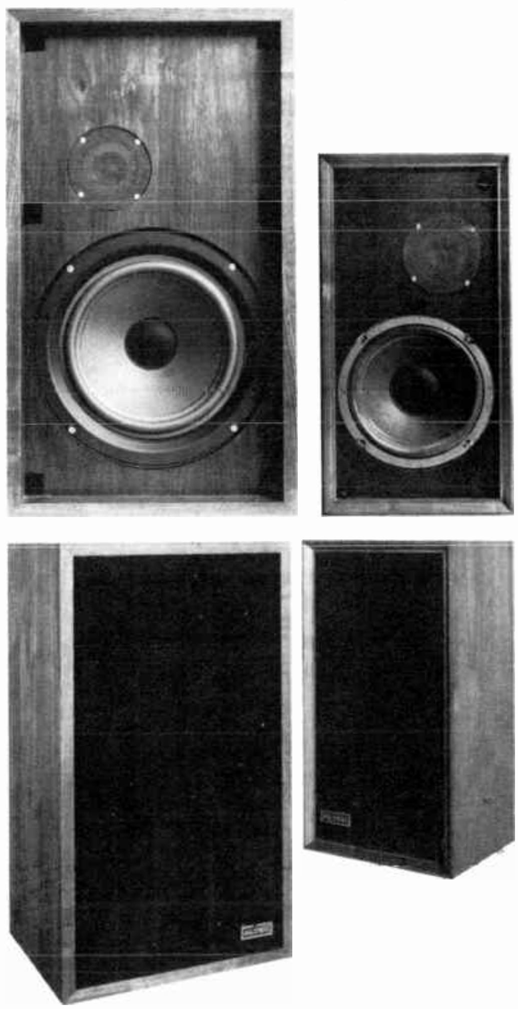
Altec knows that it takes all these criteria and more to build good studio monitor systems, and builds them accordingly. And recording professionals know Altec quality. That's why Altec is the world leader.

MINI-MONITORS

These are Altec's "Mini-Monitor" loudspeakers—the 887A Capri and the 891A Bookshelf. We call them Mini-Monitors for just one reason: their performance characteristics are amazingly similar to our actual studio systems. They deliver all of the clarity and definition of sound, the flat frequency response, the excellent transient response that recording engineers demand from a studio monitor. Yet they're specifically designed for the home. Smaller acoustic output, bookshelf dimensions, contemporary styling, and—most important—prices anyone can live with.

That's why we call them Mini-Monitors. Small wonders.

Why buy them? Because they let you hear the music the way it was first heard in the recording studio—clear and real. And if anyone should know about monitors, it's us.



Mini-Monitor I

The 891A Bookshelf. Walnut veneer enclosure and foam grille at $129. Intended primarily for those who want superior stereo—or those who can afford four-channel at this price. Economical alternative: the 891V. Same system with a walnut-grained vinyl covered enclosure and cloth grille. At $109, it saves you 20 bucks.

Altec has almost as many loudspeakers in U.S. studio use as all other brands combined.

We can prove it. Here's the latest U.S. studio data published in Billboard Magazine's 1973 International Directory of Recording Studios.

ALTEC	**514**
JBL	**256**
EV	**77**
KLH	**35**
AR	**29**
TANNOY	**28**

Throughout the world-wide recording industry, more musical esthetic decisions are made on Altec monitors than any other brand. And have been for nearly 30 years. Recording professionals listen to music through loudspeakers to earn their living. If they choose Altec, do they know something you don't?

Mini-Monitor II

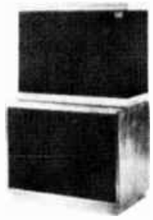
The 887A Capri. $75. Superb for smaller listening rooms. And if you want 4-channel on a budget, you got it.

ALTEC®

the sound of experience

1515 S. Manchester Ave., Anaheim, California 92803

Speakers

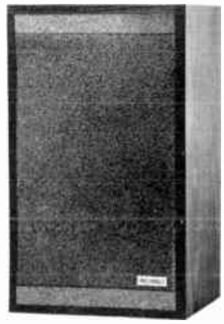


Crown ES-212

EPI 100

ESS Seven

MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp., Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
CROWN	ES 224	(2) 10	45	Acous. susp.				(24) ES pnls	22-30k	150	600	350	4	26 x 28 x 60	Wal.	Cloth, blk.	135	1165.00 ea.	Solid state protect circuitry.
	ES 212	(2) 10	45	Acous. susp.				(12) ES pnls	22-30k	150	300	375	4	26 x 21 x 42	Wal.	Cloth, blk.	110	795.00 ea	Same as above.
	ES 26	(2) 10	45	Acous. susp.				(6) ES pnls	22-30k	75	150	1500	4	24 x 12 x 27½	Wal	Cloth, blk.	70	495.00 ea	Same as above.
	ES 14	10	45	Acous. susp.				(4) ES pnls	22-30k	40	75	1500	8	18 x 12 x 27½	Wal.	Cloth, blk.	60	335.00 ea.	Same as above.
OATHAR ACOUSTICS	DA-1	(6) 5	26	*			(6) 2¼		22-22k ± 3	30	125	**	8	15 x 25 x 13¼	Wal.	Foam; brn.	57½	600.00 pr.	*103 vents; **hi-pass filter for tweeters.
OAYTON WRIGHT	XG-8 Mk II								40-16k ± 3	50	600	none	8	39⅜ x 9½ x 39⅜	blk. satin ano.	Cloth, blk.	54 ea.	1980.00 pr.	8 full rng. electro-static drives; mtc transformer; bias sply. unit; conn. cables.
OESIGN ACOUSTICS	D-12	10	47	Vent.	5	Cone	2½	Cone	30-15k ± 2	20	50	700; 1500	8	30" high w/ped.	Wal.	Cloth, blk.	45	399.00 ea.	Mult. tweet. (9): grille color opt.
	D-6	10	50	Vent.	5	Cone	2½	Cone	30-15k ± 3	20	50	700; 2000	8	16½ x 13¾ x 24½	Wal.	Cloth, blk.	30	249.00	Mult. tweet. (5): grille color opt.
OUKANE	6A425	15		w/accous. lens					40-20k		50	1000	25	32 x 18½ x 28	Wal.	Cloth, brn.	120		
OYNACO	A-50	(2) 10		Acous. susp. *			1½	Dome	35-17k ± 5	25	50	1k	8	21½ x 10 x 28	Wal.	Linen, bge.	47	189.00	*Dual chmbr
	A-35	10		Acous. susp. *			1½	Dome	38-17k ± 5	20	35	1.2k	8	12½ x 10 x 22½	Wal.	Linen, bge.	30	120.00	*As above.
	A-25	10		Resist. loaded vent			1½	Dome	44-15k ± 5	20	35	1.5k	8	11½ x 10 x 20	Wal.	Linen, bge.	24	89.00	Also in rose or teak, $99.00.
	A-10	6½		Resist. loaded vent			1½	Dome	60-15k ± 5	15	25	2.5k	8	8½ x 8 x 15	Wal.	Linen, bge.	30*	104.00*	*Pair.
EPI	50	6	55	Acous. susp.			1	Acous.	55-18k ± 3	10	35	1800	8	10 x 13 x 8	Wal.	Cloth, blk.	15	110.00	
	MT-I 75			*	4	Cone			50-10k ± 5	4	25		8	32 x 8½ x 8½		Cloth, blk.	15	120.00	*Organ Pipe Principle *Cabinet provides base from 50-200.
	100	8	45	Acous. susp.			1	Acous.	45-18k ± 3	12	50	1800	8	9 x 11 x 21	Wal.	Cloth, blk.	25	188.00	
	150	8	35	Acous. susp.			1	Acous.	35-18k ± 3	15	60	1800	8	11 x 15 x 25	Wal.	Cloth, blk.	30	278.00	
	201	8	33	Acous. susp.			1	Acous.	30 x 18K ± 3	20	100	1800	4/16	11 x 18 x 28	Wal.	Cloth, blk.	40	398.00	
	202	8	40	Acous. susp.			1	Acous.	35-18K ± 3	20	100	1800	4/16	15 x 15 x 25	Wal.	Cloth, blk.	40	438.00	
	400	6	33	Acous. susp.			1	Acous.	28-18K ± 3	30	200	1800	8	14 x 14 x 38	Wal.	Cloth, blk.	90	389.00	
	602	8 6	40	Acous. susp.			1	Acous.	35-18k ± 3	30	150	1800	4	15 x 24 x 16	Wal.	Cloth, blk.	60	498.00	
	1000	8	25	Acous. susp.			1	Acous.	22-18k ± 3	60	250	1800	8	18 x 18 x 75	Wal	Cloth, blk.	180	2000.00	
ESS	Heil AMT 1	10	30	Resis. port	2x5	*	*	*	45-24k ± 2	15			4	14½ x 14½ x 31	Wal.	Cloth, brn/blk.		299.00 ea.	*Air motion trans.
	Tempest	12	30	Resis. port			1	Dome	35-18k ± 4	20			4	15½ x 12¼ x 27	Wal.	Cloth, brn./blk.		149.00 ea.	Hi accel. drivers
	Satellite 4 System	12	25	Resis. load.	6	Cone	2¼	Cone	35-20k ± 3.5	15			8	10 w/h 7 d. 18⅝ x 20 x 17	Wal.	Cloth brn/blk		579.00 compl. sys.	Designed for 4-chan.; 4 satellites, 1 bass cube; 100 W rms amp & x-over
	Seven	9x12	40	Resis. port	5	Cone	2¼	Cone	35-20k ± 3.5	25			8	16 x 14 x 27	Wal.	Cloth brn/blk		239.00 ea.	Foam plast. oval woofer for zero cone breakup.



MODEL A-76 10" 2-WAY BOOKSHELF
LOUDSPEAKER SYSTEM

MODEL A-200 12" 4-WAY FLOOR-STANDING
LOUDSPEAKER SYSTEM

"The crescendo handling capability of the A-200 is extremely good."

—Richard C. Heyser, Audio Magazine

MODEL A-100 10" 3-WAY BOOKSHELF
LOUDSPEAKER SYSTEM

"... unusually flat toward the top for a bookshelf system in this price range."

—Bob Long, High Fidelity Magazine

IT'S UP TO YOU

What kind of sound do you want? New England? California? Funky?

With AUDIOANALYST high fidelity loudspeakers you can have it all!

AUDIOANALYST designs its loudspeakers for superior high-end dispersion, transient response, power handling, component reliability, sound accuracy through the significant frequency range (50-15, 000Hz), freedom from coloration and, most of all, astonishingly better overall *clarity* of material reproduced.

What does all this mean to you?

It means AUDIOANALYST loudspeakers provide the kind of accurate, linear sound reproduction you can build your system around.

Yes, we suggest you start with the most accurate loudspeakers available and *then* add your other components. Why choose a turntable, tape deck and amplifier of impeccable specifications and then, as an afterthought, sour it with "regional sounding" loudspeakers which, no matter how you twist the amplifier controls, will sound essentially "regional".

Start with accurate AUDIOANALYST loudspeakers and with a twist of the controls you can have New England full-range *clarity,* California *clarity* down to the biggest bass, or even hear *clarity* at its funkiest.

YOU CAN HAVE IT ALL!

6 YEAR WARRANTY

Audioanalyst

Box 262, Brookfield, Connecticut 06804

SPECIALISTS IN ENVIRONMENTAL SOUND RESEARCH

(THE PEOPLE WHO BUILT THE "ACCURATE SOUND" BANDWAGON)

Speakers

Empire Grenadier 7500MII

Equasound IIa

Fairfax

Fisher ST-465

MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp. Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
ELECTRO-MUSIC	450	15	55	Bass refl.	(2) 6x 9	Cone	1	Dome	*	*	*	800 5K	*	29½ x 19⅝ x 33	Wal.	Cloth, brn. & gold	227/ pr.	995.00/ pr.	*Input 1.5V. rms across 10K; 2 built-in 50W. rms amps; moving baffle sys. cuts standing waves.
	430	12	60	Bass refl.	3 x 9½	Horn				*	*	800	*	24⅞ x 18½ x 29⅜	Wal.	Cloth, brn.	200/ pr.	695.00/ pr.	*As above.
ELECTRO-VOICE	Inter-face: A	8/12	32	Vented piston			(2) 2½	Dome/ piston	30-18k ± 3	30		*	**	14 x 7¾ x 22	Wal.	Dbl. knit cloth blk.		400.00 pr. w/ equal.	*Elect. equalizer; **use tape monitor.
	E-V 13	8		Acous. susp.			2½	Cone	50-18k			1500	8	19 x 8½ x 10	Wal. vinyl	Cloth, brn.	15	59.95	Bal. contl.
ELITE	E-202	12	35	Acous. susp.	6	Cone	4	Flare ring	30-20k ± 4	10	45	3000; 8500	8	14¾ x 12¼ x 23¾	Wal.	Cloth brn/blk	40		
EMPIRE	9500MII	(2) 12	20	Horn load.	(2) 5	Dome	(2) 1	Dome	25-20k ± 3	10	200	450; 5000	8	30 x 20¾ x 28	Wal.	none	120	349.95	Incl. imprtd. marble top; w/wal. top, $319.95.
	7500MII	15	25	Horn load.	5	Dome	1	Dome	25-20k ± 3	10	100	450; 5000	8	20 dia. x 27	Wal.	none	90	199.95	Incl. imprtd. marble top; w/wal. top. $184.95.
	6500	12	30	Horn load.	4	Cone	2	Cone	25-18k ± 3	10	75	500; 5000	8		*	none	45	139.95	*Molded white.
	6000M	10	40	Horn load.	4	Cone	2	Cone	30-18k ± 3	10	75	500; 5000	8	18 dia. x 24	Wal.	none	60	119.95	Incl. imprtd. marble top; w/wal.. $119.95
EQUASOUND	IIa	10	30	Reflex	4½	Cone	1	Dome	30-20k ± 3	3	70	1000; 3000	4	12½ x 12½ x 41½	Wal.	Cloth, opt.	65	599.00 pr.	3½ ft. col.; colors blk., red, white; equalizer.
FAIRFAX	F2A	8	45	Bass refl.			3½	Cone	35-20k	8	25	5000	8	12 x 9 x 18	Wal.	Cloth, brn.	22	69.95	¾" board constr.
	FX100B	8	45	Duct. port			3½	Cone	32-20k	8	30	2000	8	12 x 10 x 22	Wal.	Cloth, brn.	24	89.95	Same as above.
	FX-300	10	37	Duct. port			3½	Cone	24-20k	10	40	1500	8	14 x 10¾ x 22	Wal.	Cloth, brn.	26	109.95	1" board constr.
	Wall of Sound II	(4) 8	32	Lab.	5	Cone	1 (2) 3½	Dome Cone	20-20k	15	60	750; 2000; 4000	8	23½ x 6½ x 39¼	Wal.	Cloth, blk.	60	279.95	Same as above.
	Wall of Sound I	(6) 8	30	Lab.	(2) 5	Cone	(2) 1 (2) 3½	Dome Cone	20-20k	20	100	750; 2000; 4000; 9000	6.5	30 x 6½ x 52	Wal.	Cloth, blk.	125	399.95	Same as above.
FISHER	XP7S	12	65	Acous. susp.	5	Cone	3	Cone	31-20k	10	30	600; 3000	8	13½ x 24⅛ x 12	Wal.	Sculpt. cloth, brn.	32	169.95	
	XP65S	10	68	Acous. susp.	5	Cone	3	Cone	33-20k	10	25	600; 3000	8	23 x 13 x 10	Wal.	Sculpt. cloth brn.	27	109.95	
	ST550	15	52	Acous. susp.	(2) 1½	Soft dome	(2) 2 (2) 1½	Cone Soft dome	35-20k ± 5	25	50	600; 6000; 1000	8	30 x 17 x 12¾	Wal.	Sculpt. cloth, brn.	76	349.95	Side-firing disp. domes; 2 angl. mdrng. domes; 2 angl. tweet. 530 sim. but w/o soft domes or angl. mdrng., $249.95
	ST500	12	55	Acous. susp.	1½	Soft dome	(2) 2	Cone w/ctr. dome	40-20k ± 5	25	25	600; 6000	8	26 x 15 x 12	Wal.	Sculpt. cloth. brn.	46	199.95	1 set angl. tweet.
	ST465	12	42	Acous. susp.	3½	Flare dome	1	Mylar dome	40-20k	25	50	450; 5000	5-8	24½ x 14¼ x 11¾	Wal.	Sculpt. cloth. blue	39	169.95	Butyl-edge woofer; 2 rear lev. contls.
	PL 6		45						40-18k	8	30	1000	8	29¾ x 23¾ x 2⅞	Wal.	9 patterns	21	138.00	Flat poly. diaph. w/woofer & tweet.; PL3 smaller, less pwrfl. version, $79.95.

Frazier Concerto

This newest addition to the Frazier family has the distinctive look of fine furniture. There's no mistaking the quality of sound reproduction. It's definitely Frazier. The Concerto delivers the kind of realism it takes 30 years experience to engineer. Classical sounds with concert hall clarity. Today's sounds with the excitement of a live performance.

All this in a beautiful oiled walnut cabinet with a sculptured, removable foam grille in burnt orange, brown or black. The speaker has a floor-standing base and it's perfect end table height.

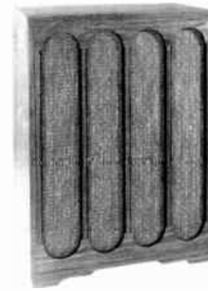
Concerto by Frazier. For people who take sound seriously.

FRAZIER
Incorporated
1930 Valley View Lane
Dallas, Texas 75234
214-241-3441

Speakers

Frazier Manhattan

GTE Sylvania AS105

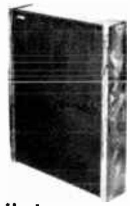


Hartley Concertmaster

MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp., Hz to kHz, ± dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
FRAZIER	Seven	12		Tuned port	(2) 4	Cones	(2)	Horns	30-18k			400; 3k	8	19 x 29 x 16	Oil wal.	Foam		310.00	
	Mark VI	12		Tuned port	8	Cone		Horn	25-17k			600; 17.5k	8	25¾ x 29 x 16⅛	Oil wal.	Cloth		350.00	
	Mark V	12		Tuned port	(2) 4	Cone		Horn	30-17k			800; 3.3k	8	14 x 25¾ x 12	Oil wal.	Cloth; brn.		235.00	
	Concerto	10		Tuned port				Horn	30-17k			15.k	8	19 x 16 x 17	Oil wal.	Foam		210.00	
	Manhattan III	10		Tuned port				Horn	35-17k			2k	8	23⅞ x 19 x 11⅞	Oil wal.	Cloth; brn.		190.00	
	Monte Carlo IV	8		Tuned port			3	Cone	50-15k				8	19 x 10¾ x 11⅞	Oil wal.	Cloth; brn.		85.00	
GTE SYLVANIA	AS85W	8	55	Acous. susp.			1½	Dome	40-20k		50	3000	8	10⅝ x 10½ x 18⅝	Wal.	Cloth, blk.	26	74.95 ea.	Mid-rng/tweeter lev swit.
	AS105W	10	50	Acous. susp.	1½	Dome	1	Dome	30-20k		65	600; 10000	8	13¾ x 11¾ x 24⅛	Wal.	Cloth, blk.	48	119.95 ea.	Mid-rng lev. swit.
	AS125B	12	42	Acous. susp.	1½	Dome	1	Dome	20-20k		100	600; 6000	8	15¾ x 12½ x 27¼	Wal.	Cloth, brn	76	169.95 ea.	3 pos. lev. swits. for mid-rng. & tweeter.
	AS225A	12	36	Acous. susp.	4½ 1½	Cone Dome	1	Dome	20-20k		150	350; 2000; 9000	8	17⅝ x 12½ x 28½	Wal.	Mld. latt. brn.	84	249.95 ea.	3 pos. lev. swits. for lower & upper mid-rng. & tweeter; remov. grille, fin. baffle.
GRACOM	P5302/SL	10		Sealed	6	Cone	(2) 1	Dome	25-25k +2 -8	17	200	250, 3k	8	12½ x 9⅞ x 23½	Wal or Rose	Metal, silver	57	381.00	12.5k rad. angle = 100°
	P4302/SL	8		Sealed	1½	Dome	1	Dome	30-25k +2 -8	15	150	800, 3k	8	18 x 7⅞ x 9¾	Wal. or Rose	Metal, silver	49	255.00	
	SM640	10		Sealed	6	Cone	1	Dome	30-25k +2 -8	13	100	650, 1.6k	8	11 x 7⅞ x 22	Wal. or Rose	Metal, Silver	43	156.00	12.5k rad. angle = 100°
	SM625	6¾		Sealed			1	Dome	45-25k +4 -8	8	40	1.7k	8	15 x 6¾ x 8½	Wal. or Rose	Metal, silver	35	98.50	Same as above.
HME	500D	8.5	50	Acous. susp			1½	Dome	50-17.5k ± 5	15	60	1,250	8	20 x 8⅛ x 12¾	Wal.	Cloth, brn.	35	109.50 ea.	Sep. hi & lo fuses.
	750B	10	46	Acous. susp.			1½	Dome	50-15k ± 5	25	70	1,250	4	23 x 9⅝ x 14	Wal.	Cloth, bm.	45	350.00 pr.	Sep. hi & lo fuses.
HARTLEY	Concert Master V	24	13	Semi inf.	10	Poly. cone	7 1	Dome Dual cone	16-25k ± 3	20	50	250; 3000; 6K	12	40½ x 29 x 18	Wal.	Cloth, blk.	150	850.00 pr.	Mag. susp.; cast alum. frames; Ident. cone mat.; heavy mags; 12dB/oct. x-over.
	Concert Master VI	24	13	Semi inf.	10	Poly. cone	7 1	Dual cone dome	16-25k ± 3	20	50	250; 3000; 6K	12	40½ x 29 x 18	Wal.	Cloth brn & gold	150	875.00 pr.	Same as above.
	Concert Master Jr.	10	28	Acous. susp.			1	Dome	30-25k ± 4	15	30	2500	8	30 x 24 x 14	Wal.	Cloth, gold & brn.	85	330.00 pair	Mag Susp.; cast alum. frame; Mediterranean styling.
	Holton A	10	28	Acous. susp.			1	Dome	30-25k ± 4	15	30	2500	8	30 x 24 x 14	Wal.	Cloth blk. on blk.	85	315.00 pair	Same as above; traditional cab.
	Holton Jr.	10	28	Acous. susp.			1	Dome	30-25k ± 4	15	30	2500	8	30 x 15 x 12	Wal.	Cloth blk. on blk	50	265.00 pair	Floor unit or lge bookshelf.
	Zodiac 74	10	30	Acous susp.			1	Dome	38-25k ± 5	10	50	2500	8	30 x 15 x 12	Wal	Foam brn. & mocha	50	128.00	Treated woofer; same cab. as Holton Jr.
	Zodiac A	10	30	Acous. susp.			1	Dome	35-25k ± 5	10	50	2500	8	30 x 24 x 14	Wal.	Brn. & mocha	85	200.00	Cab. same as Holton A.
	Zodiac C	10	30	Acous. susp.			1	Dome	35-25k ± 5	10	50	2500	8	30 x 24 x 14	Wal.	Cloth, brn. & gold	85	220.00	Cab. same as Concert Master Jr.

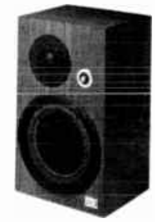
Speakers

Hegeman I



Infinity Servo-Statik I

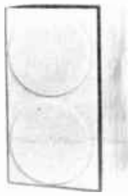
JBL L-26 Decade



JVC SX-3

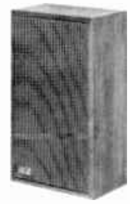
MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp., Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
HEATH	AS-38	12		Refl.			2	Cone	45-20k	5	40	2500	8	14 x 11¾ x 23	Wal.	Cloth. brn.	37	159 95K	Kit version of JBL sys
	AS-48	14		Refl.			2	Cone	40-20k	5	40	2000	8	14 x 12 x 23½	Pecan	Cloth. brn.	42	189.95K	Kit version of JBL sys
	AS-103A	12	42	Acous.	1½	Dome	¾	Dome	20-20k	25	100	575; 5000	4	14 x 11¾ x 25	Wal.	Cloth. brn./ gold	53	189.95K	Kit version of AR3a.
	AS-101	15		Refl.				Compr. driver	35-22k	5	50	800	8	27¾ x 19⅞ x 29⅝	Pecan	Cloth. blk./ gold	101	269.95K	Kit version of Altec sys.
HEGEMAN	Hegeman I	8	18	Clsd. box bffl.			1	Dome	28-20k ± 2.5	25-60	25	5k	8	11 x 8¾ x 26	Teak	Cloth. blk.	32	114 00 ea.	Drawn alum. cone; hemis. disp. pattern.
HITACHI	HS-220	8		Acous. susp.			2½	Cone	60-20k +0 -8		20	3k	8	11 x 19 x 10	Wal.	Cloth. gray	16	65.00	
	HS-420	10		Bass Refl.	5	Cone	2	Horn	45-20k +0 -8		40	2k; 8k	8	15 x 26 x 10	Wal.	Cloth. gray	33	140.00	
	HS-350	8		Bass Refl.			2	Horn	40-20k +0 -8		50	3.5k	8	14 x 23 x 10	Wal.	Cloth. gray	33	185.00	Gathered edge woofer
	HS-500	8		Bass Refl.			2¾	Horn	25-20k +0 -8		50	3k	8	14 x 24 x 14	Wal.	Cloth. brn.	48	300.00	Same as above.
IMF	Monitor III	8x12	18	Trans. line	5	Cone	2½ ¾	Dome	18-30k ± 5	60	35	375; 3.5k; 12k	8		Wal.	Cloth. blk.	140	1800.00	Dual trans. line mon. ldspkr.
	Studio IIIa	8	24	Trans. line	4	Cone	1 ¾	Dome	24-30k ± 5	60	35	375; 3.5k; 12k	8		Wal or teak	Cloth. blk.	72	800.00	Dual line spkr. for mon. or home.
	ALS 40a	(2) 8	56	Active trans. line	4	Cone	1	Dome	38-20k ± 5		35	120; 500; 3.5k	8		Wal. or teak	Cloth. blk.	50	600.00	Pat. "Active" line produces mon. qual bass
INFINITY	POS I	10		Trans. line				Cone	35-19k ± 4	15	40	1500	8	13 x 12 x 24	Vin. Wal.	Foam blk.	50	196.00 pr	
	1001	12		Trans. line				Cone	30-21k ± 4.5	20	60	1300	8	14½ x 12¼ x 25	Wal.	Cloth. blk.	65	278.00 pr.	Rear rad. tweeter
	Monitor	12		Trans. line		Dome	4"		26-32k	35	100	500; 3000	8	14 x 13 x 39	Wal.	Foam, blk.	95	858.00 pr.	Coherent rad. in wave trans. line tweeter.
	Servo-Statik I	18		Fdbk. clsd. box		ES	(4) 6" strips	ES	15-30k ± 2	100 mrng. 35 twtr.	200	100; 1500		Screens: 37 x 6½ x 28	Rose	Cloth. blk.	115	2100.00 sys	Fdbk. bass sys; ES midrng; tweeters tri amp'd; bass commode dim.: 22W x 13 D x 22 H
INNERMEDIA	Tri-Planer IIB	12	25	Acous. susp.	4½	Cone	2	PZ	30-22k ± 2½	10	40	300; 2.2k	8	16 x 8 x 32	Wal.	Plas.; blk.	50 ea.	250.00/ pr	
JBL	L100 Century	12		Bass refl.	5	Cone	1.4	Cone		2	50	1500; 6000	8	24 x 14 x 14	Wal.	Foam	55	273 00 ea.	Foam avail. in var. colors.
	L26 Decade	10		Bass refl.			1.4	Cone		2	35	2000	8	24 x 13 x 13	Oak	Stretch cloth	41	129.00 ea.	Var. colors avail. for grille
	L45 Flair	15		Bass refl.	5	Cone	1.4	Cone		1	75	1200; 7500	8	30 x 18 x 22	Wal.	Sculp. foam char. & brn.	108	426.00 ea	Opt. flr. stndng encl. avail. ($198.00)
	L200 Studio Master	15		Bass refl.				Horn		1	100	1200	8	24 x 21 x 33	Wal.	Foam	139	597.00 ea.	Comp. driver w/horn & acous. lens; var. grille colors.
JVC	SX-3	10	60	Acous. susp.			2	Soft Dome	35-20k	20	25	2000	8	12⅜ x 11' x 20½	Wal. or Spruce		29	159.95	Open baffle, wide-dir. sys.
	VS-5333	12	40		6½	Cone		Horn w/ diff.	38-20k	15		1000; 5000	8	15 x 12½ x 25½	Wal.	Cloth. brn	35	189.95	Var. lim. overload protector
	VS-5323	10	60		4	Cone		Multi cell Horn	40-20k	10	25	2000; 7000	8	12⅝ x 11¾ x 22¾	Wal.	Cloth. brn.	18	199.95 pr.	Tun. duct. port sys.
	VS-5397	8	50				2½	Cone	40-20k	8	20	4000	8	7⅞ x 7⅞ x 27½	Wal.		9	99.95/ pr	2-way; wide dir.

Speakers



JansZen Z-412

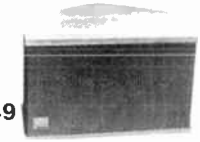
Jensen 15



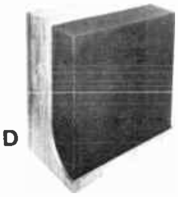
KLH 38

MANUFACTURER	MODEL	WOOFER Diameter, in.	Resonance (in system), Hz	Enclosure type	MID-RANGE Diameter, in.	Type	TWEETER Diameter, in.	Type	Overall freq. resp., Hz to kHz, ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
JANSZEN	Z-210A	10	55	Acous.			(2) 32 sq. in.	ES	45-20k ± 3	15	50	1800	6.9	12½ x 17½ x 11	Wal	Foam, blk.	30	119.95	3 pos. EST lev.
	Z-110	10	45	Acous.			16 sq. in.	ES	35-20k ± 3	20	50	1800	6.9	13¼ x 24 x 11¾	Wal.	Foam, blk.	40	129.95	Slnt bld refl. lens; vert. &/or horiz. oper.; 3 pos EST lev.
	Z-410	10	45	Acous.			(4) 64 sq. in.	ES	35-20k ± 3	20	75	1800	6.9	13¼ x 24 x 11¾	Wal.	Foam, choice	41	179.95	Frnt. pnl. cont. var. EST lev.; 5 way bind. posts; vert/hor. plcmnt.; JK 410 (kit), $149.95.
	Z-412A	12	43	Acous.			(4) 64 sq. in.	ES	33-20k ± 3	20	100	1800	4.6	14½ x 27⅛ x 11¾	Wal.	Foam, choice	44	249.95	Same as above; JK412A (kit), $179 95
	Z-412 HP	12	38	Acous			(4) 64 sq. in.	ES	30-20k ± 3	20	150	800	4.8	14½ x 24 x 11¾	Wal	Foam, blk abst	48	299.95	Hi pwr. woofer & ES version of 412A. Bi-amp fac.; JK412A HP (kit), $222.95.
	Z824	(2) 12	38	Acous.			(8) 128 sq. in.	ES	30-20k ± 3	20	300	800	8	29 x 33 x 16	Ok. oak	Foam, choice	100	650.00	Dual hi pwr. woofers; hi pwr. EST; bi-amp fac
	1300W						(4) 64 sq. in.	ES	700-20k ± 3	20	100	700	8	22 x 7¼ x 13	Wal.	Cloth, brn.	16	199.95	Mid/hi freq. EST array for use w/existing woofer.
JENSEN	16	8	100	Acous. susp.			2¾	Cone	70-13k ± 5	10	30	6000	8	14½ x 10 x 8½	Wal.	Cloth, brn.	14	39.00	Removable grille
	17	8	70	Acous. susp.			3½	Damp. dir. rad.	60-18k ± 5	10	35	1200	8	18¾ x 11 x 8¾	Wal.	Cloth, brn.	22	63.00	Remov. grille; HF bal. contl.; push-type term binding posts
	5	12	45	Acous. susp.	(2) 5		1	Dome	45-20k ± 5	10	60	500; 4000	8	26 x 15 x 13	Wal.	Cloth, brn. & blk.	52	162.00	Remov. grille; HF & MF bal. contl.; push-type binding posts.
	15	15	30	Duct. port	8		5 (2) 1	Dome	35-20k ± 5	10	100	300; 1500; 4000	8	31 x 23 x 7	Wal.	Foam, blk.	85	396.00	Remov. grille; black slate top; front mounted bal. contls.
KLH	Five	12	44	Acous. susp.	(2) 3	Cone	1¾	Cone		25		600; 2500	8	26 x 11½ x 13¼	Oil wal.	Cloth; brn.	54	199.95	2 level contls.; finish 4 sides.
	Six	12	55	Acous. susp.			1¾	Cone		10		1500	8	23½ x 12⅝	Oil wal.	Cloth; wte.	40	139.95	Tweet. contl.; finished 4 sides.
	17	10	60	Acous. susp.			1¾	Cone		10		1500	8	23½ x 9 x 11¾	Oil wal.	Cloth; wte.	27	79.95	As above.
	32	8	59	Acous. susp.			1⅞	Cone		12		1800	8	19⅜ x 7¼ x 10⅞	Oil wal.	Cloth; brn.	21	55.00	Packed 2 @ carton; $95.00/pair.
	33	10	54	CAC*			1¾	Cone		12		1500	8	23⅜ x 10¼ x 12¾	Oil wal.	Cloth; brn.	33	99.95	*Contld. acous. compliance; hi contl.; finished 4 sides.
	38	10	50	Acous. susp.			1⅞	Cone		12		1750	8	21¼ x 8¾ x 12⅛	Oil wal.	Cloth; brn. stripe	28	67.00	Packed 2 @ carton, $125.00.
	28	(3) 10	44	Acous. susp.			(3) 1½	Cone		30		1750	8	17 x 26 x 15	Oil wal.	Foam; brn.	70	299.95	Radiates in 3 directions.
	31	8	60	Acous. susp.			1⅞	Cone		10		1800	8	11 x 8¾ x 17½	Oil wal.	Foam	16	89.95 pair	Brn. or burgundy grille.
LAFAYETTE	Criterion 999	15	18	Acous.	8 5	Cone Cone	2	Cone	18-22k	15	100	400; 900; 7000	8	30 x 20 x 16	Wal.	Foam, brn.	75	199.95	3 contls. on rear.
	Criterion 888	12	19	Acous.	6 5	Cone Cone	2	Cone	19-20k	15	100	400; 900; 7000	8	24 x 15½ x 12	Wal.	Foam, brn.	50	159.95	Same as above.
	Criterion 777	10	20	Acous.	6	Cone	2	Cone	20-20k	12	80	800; 4000	8	23 x 15 x 12	Wal.	Foam, brn.	43	99.95	HF & mdrng. contls.
	Criterion 666	10	28	Acous.	6	Cone	2	Cone	28-20k	10	60	800; 5000	8	22 x 14½ x 11	Wal.	Foam, brn.	38	79.95	Md-rng. & brilliance contls.
	999	15	18	Acous. susp.	8; 5	Cone; cone	2	Dome	18-22k	15	100	400; 900; 7k	8	30 x 20 x 16	Oil wal.	Foam; char. brn.	75	199.95	12-lb. mag. & 2½-in. voice coil woofer; 3 contls.
	888	12	19	Acous. susp.	6; 5	Cone; cone	2	Dome	19-20k	15	100	400; 900; 7k	8	24 x 15½ x 12	Wal. oil	Foam; char. brn.	50	159.95	As above.
	777	10	20	Acous. susp.	6	Cone	2	Dome	20-20k	12	80	800; 4k	8	23 x 15 x 12	Wal.	Foam; char. brn.	43	99.95	8-lb. mag. *2-in. voice coil woofer; mid & hi contls.
	666	10	28	Acous. susp.	6	Cone	2	Dome	28-20k	10	60	800; 5k	8	22 x 14½ x 11	Oil wal.	Foam; char. brn.	38	79.95	6-lb. mag. & 2-in. voice coil woofer; mid & hi contls.

Speakers



LDL 749



Magnum Opus 24-D

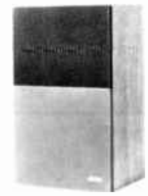
Marantz Imperial 8

MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp., Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
LINEAR DESIGN LABS	749A	(9) 4½		*						30	300		8	19 x 12½ x 12	Wal.	Cloth; brn.	43	330.00/ pr.	*Front & rear radiating: bases. $60; equal., $90.
	749A	*	*		*	*	*	*		30	300	n/a	8	19 x 12½ x 12	Wal.	Cloth, dk. brn.	43	333.00 pr.	* 9 full-rng. 4½" drvrs. (cone spkrs.) trumpet bases, $60./pr; eqlzr. avail, $90.00.
LWE	I-B	15		Elect. susp.	6	Cone	3	Horn	20-20k ± 5	10	80	2000; 4500	4	16 x 19 x 23⅞	Wal.	Cloth, brn.	73	345.00	Rm. gain, MF-HF contls.; built-in base. inv. feed-back sys.
	II-B	(2) 15		Elect. susp.	(2) 6	Cone	3	Horn	20-20k ± 5	25	130	2000; 4500	4	34 x 24 x 16	Wal	Cloth brn.	121	600.00	Rm gain, MF HF contls inv feedback sys
	III-A	12		Elect. susp.	6	Cone	2	Cone	25-17.5k ± 5	6	40	2000; 3000	4	14 x 12 x 23⅞	Wal	Cloth, brn	43	225 00	Same as above
	VI-A	8	85	Acous. susp.			2	Cone	35-17.5k ± 5	5	25	3000	8	18 x 11 x 9	Wal.	Cloth, brn.	31	80 00	Inv feedbk. sys.
	IX	10		Elect. susp.			2	Cone	30-17.5k ± 5	6	30	3000	4	12½ x 9 x 21½	Wal	Cloth, brn	42	120.00	Rm gain. HF contl., inv fdbk sys.
	X	12		Elect. susp.	6	Cone	3	Horn	25-20k ± 5	6	40	2000; 3000	4	14 x 12 x 23⅞	Wal.	Pre-formed color choice	45	275 00	Rm gain. MF-HF contls.; inv. fdbk sys.
H.J. LEAK (ERCONA)	Sandwich 600	13	19	Acous. susp.	3½	Cone	2	Dome	40-20k	4	70		8	25½ x 14⅞ x 12⅞	Wal.	Cloth, brn.	55	295 00	Piston action sand. cone; 3 way sys.
LOUDSPEAKER DESIGN	1	8		Acous. susp			(2) 3	Cone	*	15-20	**	500	8	15 x 23 x 11	Wal.	Cloth, blk.	39	218.00	Treb bal contl.; fuse; *Info avail. on req. **Pwr. hand suit for high pwrd. amps.
	2	10		Acous. susp.			(4) 3	Cone	*	15-20	**	450	8	26 x 19 x 15¾	Wal.	Cloth blk.	56	398.00	Same as above
	3	12		Acous. susp.			(8) 3	Cone	*	15-20	**	400	8	19 x 12 x 32	Wal	Cloth, blk	59	598 00	Same as above.
	4	15		Acous. susp.			(16) 3	Cone	*	15-20	**	350	8	28 x 48 x 14	Wal.	Cloth, blk.	74	1298 00	Same as above.
MAGITRAN	D560									5	28		8	29⅝ x 23⅝ x 2	Wal.	13 patterns	13	79 95	Flat sound panel; polyplanar, 8 way design
MAGNAVOX	Max 15	15	43	Acous. susp.	2	Dome	2	Phen. ring cone	20-20k	10	100	1500; 4500	8	20¼ x 15¼ x 29	Dil. Wal.	Foam blk.	55	199.95 ea	Remov. grille w/ frnt mntd mdrng. & twtr contls.
	Max 12	12	45	Acous. susp.	2	Dome	e	Phen. ring cone	25-20k	10	75	1500; 4500	8	15¾ x 13¾ x 25¾	Dil. Wal.	Foam, blk.	40	159.95 ea	Same as above.
	Max 10	10	47	Acous. susp.	2	Dome	2	Phen. ring cone	30-20k	10	50	1500; 4500	8	15¼ x 12½ x 23½	Dil Wal	Foam. blk.	35	119 95 ea	Same as above
	SD2520	12	55	Open back			(3) 3½	Cone	40-17.5k		50		8	18 x 3 x 25½		Cloth, brn.	22	79.95 ea	Floor or wall; step lev. contl; bi-dir.; angled twtrs.
MAGNUM OPUS	Opus 24	(4) 12	20		(4) 5	Cone	(8) 1¼ (8) 3	Dome Cone	20-20k ± 4	20	200	750; 3000	4/8	36 x 18½ x 32	Wal.	Cloth. brn/blk	200	1100.00 ea.	
	Opus 12	12 & 10	20		5	Cone	(3) 1¼ (6) 3	Dome Cone	20-20k ± 5	30	100	750; 3000	8	18 x 18½ x 32	Wal.	Cloth. brn/blk	105	550.00 ea.	
	Opus 7	(2) 10	25		5	Cone	(2) 1¼ (2) 3	Dome Cone	25-20k ± 5	30	100	750; 3000	4	15¼ x 13¾ x 28	Wal	Cloth, brn/blk	65	289.95 ea.	
	Opus 4	(2) 10	30		5	Cone	1¼	Dome	30 20k ± 5	30	100	750; 3000	8	15¼ x 13¾ x 28	Wal	Cloth brn/blk	63	249.95 ea.	
	Opus 2	10	33	Acous. susp.			(2) 1¼ (2) 3	Dome Cone	33-20k ± 5	20	50	2200	8	14½ x 11 x 24	Wal.	Cloth. brn/blk	44	149.95 ea.	
MARANTZ	Imperial 9	(2) 10	30	Duct. port	(4) 3½	Cone	(2) 1¾	Cone	35-17k ± 1.5	6-10	100	700; 6500	8	24 x 18 x 30½	Wal	Foam, brn	90	449.00	2-pos var. contl x over
	Imperial 8	12	35	Duct. port	(3) 3½	Cone	(2) 1¾	Cone	40-18k ± 2	6-10	75	1000 6500	8	18¾ x 14½ x 27	Wal	Foam. brn	65	299.00	2 pos mid & hi. swit. elect./acous x-over
	Imperial 7	12	42	Duct. port	3½	Cone	1¾	Cone	40-18k ± 5	6-10	50	2000; 7000	8	14¼ x 11½ x 25½	Wal.	Foam. brn	45	179.00 ea	Same as above
	Imperial 6G	10	48	Duct. port			1¾	Cone	40-18k ± 5	6-10	50	2100	8	14¼ x 11½ x 25½	Wal.	Foam, brn.	40	139.00 ea.	Elect./acous x-over; Hi freq swit; Imp 6 same w/brn cloth grille cvr, $119 ea
	Imperial 5G	8	55	Duct. port			1¾	Cone	45-16k ± 5	6-10	15	2000	8	12 x 9½ x 23	Wal	Foam. brn.	23½	99.00 ea	Same as above Imp. 5 sim w/brn. cloth grille cov., $79 ea.
	Imperial 4G	8	60	Duct. port			1¾	Cone	60-15k ± 5	6-10	15	5000	8	11¼ x 8½ x 19⅛	Wal. vin	Foam, brn	20	59.00 ea.	Elect./acous. x-over.

Speakers

Maximus M-400

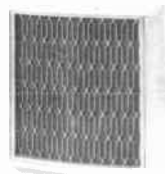
Ohm A



Pioneer R700

Quadraflex Q55

MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp., Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
MARTIN	Micro Max	8	57	Inf.			2	Dome		10	40	2K	8	11 x 10 x 18	Wal.	Cloth. brn.	20	79.00	
	Super Max	10	55	Inf.			2	Dome		15	45	2K	8	12½ x 10 x 21½	Wal.	Cloth brn	30	109.00	
	Lab Mk II	10	55	Inf.	4	Cone	2	Dome		20	50	1K; 5K	8	12½ x 10 x 21½	Wal	Cloth, brn	32	139.00	
	Crescendo	12	50	Inf.	4	Cone	2	Dome		25	60	1K; 5K	8	15 x 12 x 25½	Wal	Cloth. brn	45	199.00	
MAXIMUS	M-400	12		Vent. port	4	Cone	(2) 3	PZ						14½ x 11 x 25½			55	200 00	
	M-300	12		Vent. port	4	Cone	3	Cone						14 x 12 x 24			53	150 00	
	M-200	10		Vent. port	4	Cone	3	Cone						12½ x 9½ x 22			25	120 00	
QUADRAFLEX	66	12	'45	Acous. susp	6	Cone	3	Cone	32-20k ± 5	20	40	500: 5000	8	25⅝ x 16 x 13¾	Wal	Cloth dk. brn.		139.95	
	55	10	52	Acous. susp.	4½	Cone	3	Cone	40-20k ± 5	14	25	650; 6000	8	24 x 15 x 11¾	Wal.	Cloth. dk brn		99.95	
	44		58	Acous. susp.			3½	Cone	45-20k 5	12	25	750	8	23 x 13½ x 11	Wal.	Cloth. dk brn.		69.95	
OHM ACOUSTICS	A	18	27	Acous.					25-20k ± 4	150	200	none	8	22½ x 22½ x 48	Wal or rose	Cloth. brn.	200	1550.00	Single coherent sound: omni driver
	B	12	37	Acous.			1	Dome	35-18k ± 4	50	75	1700	8	26 x 10¾ x 15	Wal	Cloth. brn.	50	400 00	Phase-linear series type network
	C	10	43	Acous.			1	Dome	40-18k ± 4	40	75	1700	8	14 x 25 x 9¾	Wal.	Cloth, brn	45	320.00	Same as above.
	D	10	50	Slot. resist. load			3	Cone	40-16k ± 4	20	75	1700	8	14 x 25 x 8	Wal. or Vinyl	Cloth brn	40	230.00	Phenolic ring surround tweeter
	E	8	50	Acous.			3	Cone	48-16k ± 4	10	40	1700	8	11½ x 21½ x 7¼	Vinyl wal.	Cloth brn.	20	150.00	Same as above.
	F	12	35	Acous.					33-20k ± 4	50	160	none	4	17¾ x 17¾ x 43½	Wal. or Rose	Cloth brn.	125	800.00	Omnidirectional driver
OLSON	SS-82	12	48	Acous. susp.	12	Horn	6	Horn	20-30k ± 1	10	60	800; 5000	8	14⅜ x 11¾ x 23⅝	Wal.	Cloth. brn	42	329.98 ea.	3 way.
	SS-72	12	48	Acous. susp.	5	Cone	2½	Cone	25-20k ± 1	10	60	800; 5000	8	14⅜ x 11¾ x 23⅝	Wal.	Cloth brn.	40	249.98 ea.	3-way.
	SS-175	12	58	Acous. susp.	5	Cone	2½ 1¼	Cone spr. twtr	20-27k ± 1.5	10	50	700; 6000; 12k	8	14¼ x 11½ x 25½	Wal.	Fretwork wood	40	177.98 ea.	4 way.
ONKYO	Radian III	(2) 6½	90	Bass refl.			(2) 3"	Cone	60-20k	10	15	1.5k	8	9½ x 10 x 33½	Wal	Cloth. beige	27	119 95	
	15	10	60	Acous.	1⅜	Dome	1	Dome	30-20k	10	20	1K: 7K	8	11⅜ x 12⅜ x 23	Wal	Cloth. brn.	33	129.95	
	20	12	60	Acous.	2	Dome	1	Dome	35-20k	10	25	700; 7k	8	13½ x 11⅝ x 23¼	Wal	Cloth, beige	40	199.95	
	25	14	53	Acous.	2	Dome	1	Dome	30-20k	10	30	700; 7k	8	14¾ x 11⅝ x 25½	Wal.	Cloth. beige	54½	249.95	
PIONEER	R-700	12		Bass ref.		Horn	Multi cell	Horn	35-20k		75	700; 14k	8	14 x 13 x 26	Wal.	Cloth. brn. & blk.	50	229.95	2-color remov. grilles; horn type mid/high range spkr.
	R-500	10		Bass ref	5	Cone	3	Horn	35-20k		60	650; 6k	8	13 x 12 x 24	Wal	Cloth. blue & blk.	38	159 95	2 color remov. grilles.
	R-300	10		Bass ref.			3	Horn	45-20k		40	4200	8	13 x 10 x 22	Wal	Cloth. red &	26	119.95	Same as above.



Radio Shack Optimus 7

Superscope S-310

Tannoy Mallorcan

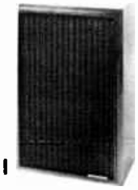
Speakers

MANUFACTURER	MODEL	WOOFER			MID-RANGE		TWEETER												
		Diameter, in.	Resonance (in system), Hz	Enclosure type	Diameter, in.	Type	Diameter, in.	Type	Overall freq. resp., Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
RTR	280DR	(4) 10	*	Inf. baf			(6) 2	Cone	22-18.5k - 4	25	100	3500	8	16½ x 16½ x 39	Wal	Cloth blk	95	280 00	*Acous fdbk crt. (pat.): total disp all freq.
	180D	(2) 10	49	Inf. baf			(4) 2	Cone	28-18.5k - 4	30	60	3500	8	14 x 14 x 33½	Wal.	Cloth blk	65	180 00	One planar resist. load woofer; total disp. hi freq.
	HPR-12	12	*	Rear load.	5	Cone	2	Cone	35-18.5k - 4	20	60	1.5k: 10k	8	14½ x 13 x 36	Wal	Cloth blk.	65	169.00	*Helmholz driven passive rad.
	ESR-6						(6) 3 x 5¾	ES	1.5-30k	30	60	1500	8	14½ x 12 x 14½	Wal	Cloth. blk	25	149 00	ES add on tweeter: built in x overs & lev contls for both tw. & wfr sys
RADFORD (AUDIONICS)	TL-30	7	45	Trans line			1	Dome	45-16k - 5	10	30	2500	8	34 x 10 x 10	Rose	Cloth blk	48	149 95	
	TL 50 A	8	45	Trans line	5	Cone	1	Dome	38-25k - 4	30	70	500 3500	8	44 x 12 x 12	Rose	Cloth blk.	80	220 00	Same HF unit as TL-90.
	TL-90	10	42	Trans. line	4	Cone	1	Dome	35-25k - 3 5	25	70	500: 4200	8	44 x 13½ x 14½	Rose	Cloth blk	98	370 00	All Radford drivers & x-over
RADIO SHACK	Optimus 7	12		Acous susp.	6½		2½	Cone	20-25k 3			850. 5k	8	24 x 15¾ x 27	Wal.	Brn	55	198 50	Metal & cloth grillework
	Optimus 5	12		Acous susp			(3) 4		20-20k - 3		100	800: 5k	8	14 x 11½ x 25	Wal.	Beige	35	109.50	
	MC-1000	8		Acous susp.			3						8	11¼ x 8¼ x 17¼	Wal	Beige	20	55 00	
	MC-500	5		Acous. susp.			2		75-20k - 3				8	11¾ x 9 x 15½	Wal	Beige	10	35.00	
SOUNDCRAFTSMEN	SC-12ES	12		Acous susp	5	Cone	144 sq in	ES	20-30k	20	150	500: 1000. 15k	8	18 x 14 x 28	Wal	Cloth, red & blk.	76	399 50	Reflectrostatic · back-wave emission dblr ; equal contl pnl.
	Lancer SC-6	12		Refl				Horn	18-22k	10	60	1k: 3k	8	16 x 14⅛ x 27	Wal.	Cloth red	57	249 50	4-way; cont var hi freq contl.
	Lancer SC-3X	12		*				Horn	26-22k	10	60	1k. 3k	8	15¾ x 12½ x 23½	Wal	Cloth. red	45	199 50	3-way; *ducted port reflex resist. loaded
SUPERSCOPE	S-26	6½	80 - 10	Port			3	Cone	100 16 5k	1 5	10	5k	8	11½ x 6¼ x 19⅛	Wal.	Vacuum form cloth, brn.	9½	59 95 pr.	
	S-212	12	35	Port			2	Cone	45-18k - 5	4	30	2k	8	14½ x 11 x 23½	Wal.	Vacuum form cloth, brn.	40	99.95	
	S-310	10	26	Port	3½	Cone	2	Cone	40-18k - 5	5	30	2k: 7k	8	14½ x 11 x 24½	Wal.	Vacuum form cloth, brn.	43	139.95	2-pos. high freq. contl.
TANNOY	Mallorcan	12	64	Refl			2½	Exp horn	45-20k	20	30	1000	8	23½ x 14⅝ x 11½	Wal.	Crvd. wood with cloth	45	255.00 each	Dyn. & freq. bal.; cab. finished 4 sides.
TELEX	TX 10A	10	50	Acous susp	5¼	Cone	3	Cone	20-20k - 6	5	45	1200: 3500	8	25 x 12 x 11½	Wal	Cane	62	99.95	

Speakers

Rectilinear III Lowboy

Rogersound RS28



SAE Mk XII

MANUFACTURER	MODEL	WOOFER Diameter, in.	WOOFER Resonance (in system), Hz	WOOFER Enclosure type	MID-RANGE Diameter, in.	MID-RANGE Type	TWEETER Diameter, in.	TWEETER Type	Overall freq. resp., Hz to kHz ± ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont)	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
RECTILINEAR	III Lowboy	12	40	duct. port	5	Cone	(2) 2½ (2) 2	Cone	22-18.5k ± 4	20	100	500; 8k 11k	8	22 x 12¼ x 28	Wal.	fret-work	70	299.00	
	III	12	40	duct. port	5	Cone	(2) 2½ (2) 2	Cone	22-18.5k ± 4	20	100	500; 8k; 11k	8	18 x 12 x 35	Wal	Cloth. brn	70	299.00	
	XII	10	45	duct. port	5	Cone	2½	Cone	35-18k ± 4	15	85	350; 7.5k	8	14 x 10¾ x 25	Wal.	Cloth. brn.	40	149.00	
	Mini III	8	50	Acous. susp.	5	Cone	2	Cone	50-18k ± 4	20	70	400; 8k	4	12 x 9½ x 19	Wal	Cloth. brn	25	109.00	
	XIa	10	45	Duct. port			3½	Cone	45-18k ± 4	10	50	1k	8	12 x 23 x 10½	Wal.	Cloth. tan	31	89.00	Fretwork, grille, opt $10 each
ROGERSOUND	RSL 28	8	67	Acous. susp			2½	Cone	50-19k ± 5	15	30+	1800	8	18 x 9⅜ x 8½	Wal.	Cloth. blk.	15	39.95 ea.	LC x-over.
	RSL Mini Monitor	8	62	Bass refl.			2½	Cone	40-19k ± 5	5	45	1800	8	22 x 12¼ x 10¼	Wal.	Opt.	29	69.95 ea.	
	RSL Studio Monitor	12	52	Bass refl.	5	Cone	2x6	Horn	40-15k ± 3	7	80	800; 5000	8	25 x 14½ x 12	Util.	Opt.	45	105 00 ea.	Avail. in wal. in var. colors, $130 00.
	RSL Max	12	37	Abs. line			(2) 2½	Cone	30-19k ± 3	15	75	1600	8	32 x 18 x 12	Wal.	Cloth. blk.	55	160.00 ea	Amb tweeter: tight bass & full imp.
	RSL 512	(2) 12	30	Bass refl.	3x9	Horn & Drive	(2) 2x6	Horn	20-18k ± 4	8	100	800; 500	4	36 x 28 x 18	Util.	Cloth. blk.	100	235.00	W/wal. encl @ 165 lbs., $275.00.
ROLA-CELESTION (HERVIC)	66	12		Acous. susp	2½	Dome	2	Dome	16-40k ± 5	10	80	500; 5000	4-8	15 x 11½ x 40	Wal.	Cloth. grey	62	486.00	Acous. bass rad.
	44	12		Acous. susp.	6	Cone	2	Dome	30-30k ± 3	6	44	500; 5000	4-8	14½ x 10 x 30	Wal.	Cloth. brn.	40	272 00	
	25	12		Acous. susp.	(2) 2	Dome	2	Dome	20-40k ± 5	5	60	3000; 9000	4-8	14 x 11 x 32	Wal.	Cloth. brn.	48	332 65	Same as above.
	15	8		Acous. susp.			2	Dome	30-15k	4	30	3000	4-8	9½ x 9¼ x 21	Wal.	Cloth. brn.	15	141 95	Same as above.
SAE	Mk XIV	12.3	18	Acous. susp.	(2) 5	Cone	(6)	ES		100 rms	*	120; 240; 480; 1440	8	24 x 18 x 42¾	Wal. or Rose	Ebony	250 pr.	1650.00 Wal. 1850.00 Rose pr.	*No limit; elect. prot. against d.c. & oscillations.
	Mk XII	12.3	18	Acous. susp.	5	Cone	(3)	ES		60	*	120; 240; 480; 1440	8	17 x 12¼ x 27	Wal. or Rose	Ebony	175 pr.	950.00 Wal. 1000.00 Rose pr.	*As above; adj. x-overs; hand rub ebony base, $30 00.
	Mk XI	12	18	Inf. baf.	(2) 5	Cone	(4) 2.5	Dome		30	300	120; 480; 1440	4	17 x 12¼ x 27	Wal.	Ebony	110 pr.	550.00 pr.	Ebony base, $300 00; hi freq. contl -7dB to +5 dB; mid. contl. -16dB to +4dB, fused
	Mk X	12	18	Inf. baf.	5	Cone	(2) 2.5	Dome		20	200	480; 1440	8	14½ x 12 x 25	Wal.	Ebony	98 pr.	400.00 pr.	Same as above.
SANSUI	SP-3500	14		Bass refl.	(2) 4 4x2	Cone Horn	(2) 2	Horn	25-20k		100	700; 2000; 6500	8	17¾ x 11½ x 25¼	Wal.	*	58.1	249.95	Multidir.; *Kumiko grille.
	SP 2500	12		Bass refl	(2) 5	Cone	(2) 5	Horn	30-20k		80	700; 6500	8	16¾ x 10⅞ x 25¼	Wal.	*	51.3	209.95	Same as above.
	SP 1700	12		Bass refl.	(2) 4	Cone	(2) 2	Cone	30-20k		70	700; 6500	8	16¾ x 10⅞ x 25¼	Wal.	*	46.7	189.95	Same as above
	AP-1200	10		Bass refl.	(2) 4	Cone	(2) 2	Cone	35-20k		60	700; 6500	8	15⅞ x 10⅞ x 23⅞	Wal.	*	42.7	169.95	Same as above.
SCHOBER	LSS-10A	12	32	Refl.	8	Cone		*	30-18k **	2	40	250; 3500	8	24 x 16 x 34	Wal.	Cane bge.	60	243.20 (kit)	*Horn tweet opt.; **w/horn
SHERWOOD	Wood-stock	8	40	Acous. susp			3½	Omni polar	40-18k	5	25	4000	8	18 x 11 x 9	Wal.	Cloth. brn	35	59.95	
SOLAR	1000	10		Acous susp.			(2) 3½	Cone	35-22k	25	60	2600	8	12 x 14½ x 24⅜	Wal.	Dbl. knit, chc.	42		Cont. var hi freq. contl.; opt. floor stand.
	100	12			4	Cone	4	Horn	35-17.5k	5	40	2500	8	12 x 14½ x 24⅜	Wal.	Dbl. knit, chc.	40		Resettable push-button crt brkr.

what is a JR-200M to an audiophile?

Sound is to the audiophile what salt is to pepper, diamond's to a girl, or an oasis to a desert. Each has a most important relationship. That's why an audiophile, a real sound expert prefers TRUSONIC JR-200M's magnificent studio quality reproduction. Clean, undistorted sound from this 12″ 3-way speaker system again demonstrates the incomparable electronic technology of TRUSONIC. THE PRICE? You can pay twice as much, but you won't find a better studio quality system. Write TRUSONIC for complete information on JR-200M.

Just a few of JR-200M's impressive engineering features are 12″ woofer for distortion-free low frequency bass performance 5″ mid-range speaker for clean voice and horn reproduction, and a horn-loaded dome radiator tweeter for high frequency. A carefully engineered crossover network, plus individual level controls to adjust midrange and tweeter sound independently, are included. Size is: 24″ x 15¼″ x 12″ deep. Finish: Hand-rubbed oil finish, Walnut veneer.

NET $149.95

TRUSONIC
1100 E. Franklin Street
Huntington, Indiana 46750

Export: Morhan Exporting Co.
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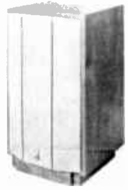
Plainview, N.Y. 11803

Speakers



Venturi Formula 6

Utah MP2000

Yamaha NS-550

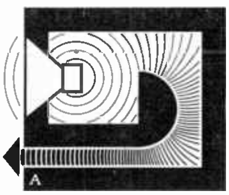
		WOOFER			MID-RANGE		TWEETER												
MANUFACTURER	MODEL	Diameter, in.	Resonance (in system), Hz	Enclosure type	Diameter, in.	Type	Diameter, in.	Type	Overall freq. resp., Hz to kHz ? dB	Ampl. pwr. for avg. room, W	Pwr. handling capacity (RMS cont), W	Crossover frequency (ies), Hz	Impedance, ohms	Enclosure dimensions, W x D x H, in.	Wood finish	Grille material color	Weight, lbs.	Price (per pair?)	SPECIAL FEATURES
TRANSDUCTION	LTD-1	12		Trans line	5 .	Cone	¾	Dome	28-20k ~ 3.5	10	60	700: 5000	4-8	15 x 13¼ x 30	Wal.	Cloth, blk.	60	250.00 ea.	Plast. lam. drive sys.: trans. line midrng.; synth. damping mat.
TRUSONIC	JR-100M	10	60	Tuned port	5	Cone	1	Dome w/horn	30-20k	10	40	1500: 5000	8	14 x 23 x 10	Wal.	Foam, chc.	33	119.95 ea.	Sld. chassis M-R; treb. lev. contl.
	JR-200M	12	68	Tuned port	5	Cone	1	Dome w/horn	20-20k	10	50	2500; 7000	8	15¼ x 24 x 12	Wal.	Foam, chc.	46	149.95 ea.	Sealed M-R; sep. M-R & twtr. cont.
VENTURI (BIC)	Formula 2	8		*		**	2	Dome	30-23k	7	75	1500; 15k	6	12 x 11½ x 19¾	Wal.	Foam, choice	27	98.00	*Venturi encl.; **Biconex horn.
	Formula 4	10		*		**	2	Dome	25-23k	5	100	1500; 15k	6	13¼ x 13 x 25	Wal.	Foam, choice	40	136.00	Same as above.
	Formula 6	12		*		**?	(2) 2	Dome	20-23k	2	125	750; 1500; 15k	6	15¾ x 14¾ x 26¼	Wal.	Foam, choice	54	239.00	Same as above.
UTAH	AS-2AX	8	100	Acous.			3½	Cone	45-17.5k	10	12	5000	8	11¼ x 17¼ x 8¼	Wal.	Cloth, brn.	20	49.95 ea.	Sld. back twtr.: oil. wal veneer.
	MP-2000	12	60	Duct. port	5	Cone	1	Dome w/horn	30-20k	10	30	2500: 5000	8	15¼ x 24 x 12	Wal.	Foam, chc.	42	139.95 ea.	Sld. chassis M-R; treb. lev. contl.
	HS4-B	12	60	Duct. port	3½ x 8	Compr. drvr. w/horn	1	Dome w/horn	25-19.5k	5	45	2500; 5000	8	15 x 25¾ x 14	Wal.	Foam, chc.	49	149.95 ea.	M-R compr. unit w/ellip. horn; treb. lev. contl.
YAMAHA	NS-570	15x20	50	Clsd.			2	Horn			45	5000	8	17 x 13 x 44	*	Cloth, blk.	88	500.00	*White poly/Rose; var. tweeter lev.; remov. grille.
	NS-550	12x18	55	Clsd.			2	Dome			30	5000	8	16 x 12 x 40	*	Cloth, blk.	71	400.00	*White poly/Wal.; Same as above.

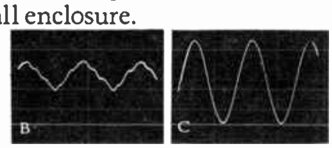
Now BIC VENTURI™ puts to rest some of the fables, fairytales, folklore, hearsay and humbug about speakers.

Fable

Extended bass with low distortion requires a big cabinet.

Some conventional designs are relatively efficient, but are large. Others are small, capable of good bass response, but extremely inefficient. The principle of the BIC VENTURI systems (pat. pend.) transforms air motion velocity within the enclosure to realize amplified magnitudes of bass energy at the BIC VENTURI coupled duct as much as 140 times that normally derived from a woofer (Fig. A). And the filtering action achieves phenomenally pure signal (Scope photos B & C). Result: pure extended bass from a small enclosure.





B—Shows output of low frequency driver when driven at a freq. of 22 Hz. Sound pressure reading, 90 dB. Note poor waveform.

C—Output of venturi coupled duct, (under the same conditions as Fig B.) Sound pressure reading 111.5 dB, (140 times more output than Fig. B.) Note sinusoidal (nondistorted) appearance.

Fairytale

It's okay for midrange speakers to cross over to a tweeter at any frequency.

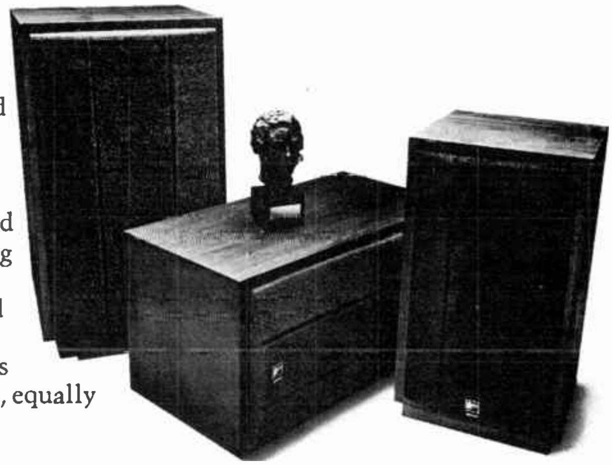
Midrange speakers cover from about 800 Hz to 6000 Hz. However, the ear is most sensitive to midrange frequencies. Distortion created in this range from crossover network action reduces articulation and musical definition. BIC VENTURI BICONEX horn (pat.pend.) was designed to match the high efficiency of the bass section and operates smoothly all the way up to 15,000 Hz, without interruption. A newly designed super tweeter extends response to 23,000 Hz, preserving the original sonic balance and musical timbre of the instruments originating in the lower frequencies.

BICONEX™ horn

Folklore

Wide dispersion only in one plane is sufficient.

Conventional horns suffer from musical coloration and are limited to wide-angle dispersion in one plane. Since speakers can be positioned horizontally or vertically, you can miss those frequencies so necessary for musical accuracy. Metallic coloration is eliminated in the BICONEX horn by making it of a special inert substance. The combination of conical and exponential horn flares with a square diffraction mouth results in measurably wider dispersion, equally in all planes.



Hearsay

A speaker can't achieve high efficiency with high power handling in a small cabinet..

It can't, if its design is governed by such limiting factors as a soft-suspension, limited cone excursion capability, trapped air masses, etc. Freed from these limitations by the unique venturi action, BIC VENTURI speakers use rugged drivers capable of great excursion and equipped with voice coil assemblies that handle high power without "bottoming" or danger of destruction. The combination of increased efficiency and high power handling expands the useful dynamic range of your music system. Loud musical passages are reproduced faithfully, without strain; quieter moments, effortlessly.

Humbug

You can't retain balanced tonal response at all listening levels.

We hear far less of the bass and treble ranges at moderate to low listening levels than at very loud levels. Amplifier "loudness" or "contour" switches are fixed rate devices which in practice are *defeated* by the differences in speaker efficiency. The solution: Dynamic Tonal Compensation.™ This circuit (patents pending) adjusts speaker response as its sound pressure output changes with amplifier volume control settings. You hear aurally "flat" musical reproduction at background, average, or ear-shattering discoteque levels—automatically.

A system for every requirement

FORMULA 2. The most sensitive, highest power handling speaker system of its size (19¾ x 12 x 11½)." Heavy duty 8" woofer, BICONEX mid range, super tweeter. Use with amplifiers rated from 15 watts to as much as 75 watts RMS per channel. Response: 30 Hz to 23,000 Hz. Dispersion: 120° x 120°. $98 each

FORMULA 4. Extends pure bass to 25 Hz. Has 10" woofer, BICONEX midrange, super tweeter. Even greater efficiency and will handle amplifiers rated up to 100 watts. Dispersion: 120° x 120°. Size: 25 x 13¼ x 13." $136 each.

FORMULA 6. Reaches very limits of bass and treble perception (20 to 23,000 Hz). Six elements: 12" woofer complemented by 5" cone for upper bass/lower midrange; pair of BICONEX horns and pair of super tweeter angularly positioned to increase high frequency dispersion (160° x 160°). Size: 26¼ x 15¾ x 14¾." $239 each.

Sturdily constructed enclosures. Removable grilles in choice of 6 colors. Optional bases for floor standing placement. Write for brochure AH FH-74

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BIC VENTURI™

The Loudspeaker As A Spherical Sound Source

***Winslow N. Burhoe**

It has long been known to physicists and audiophiles that the ideal loudspeaker would radiate sound equally in all directions, at all frequencies without any distortion. The usual visual image which is called to mind is that of a small pulsating sphere, perhaps the size of a tennis ball. As this imaginary ball expands and contracts, it transmits a pressure wave to the air which then expands as a spherical wave of sound. Unlike wave motion in strings or on the surface of water, sound waves are three dimensional.

The term "omnidirectional" applies to the three dimensional spherical wave pattern this ideal sound source would generate. This term has been abused in recent years by being applied to speakers that do not technically qualify as omnidirectional. Some have been omnidirectional only over a very narrow frequency bandwidth and most are not omnidirectional at any frequency. As a consequence, there has been little industry or consumer excitement over speakers which have been introduced and labelled "omnidirectional".

It is essential for true and valid omnidirectionality that two conditions be met: (1) Omnidirectionality at all frequencies, and (2) equal energy radiation in *all* directions—up, down, left, right, forward and backward. Omnidirectionality is NOT a substitute for other high fidelity specifications; it cannot compensate for poor frequency response or high distortion. It is of no advantage unless applied to wide frequency bandwidth. However, when the traditional high fidelity values of flat frequency response, low distortion, and wide bandwidth are embodied in a truly omnidirectional speaker, a major improvement in sound reproduction is achieved; the close approximation of the mythical ideal speaker, a Spherical Sound Source.

Like all other historical advances in the art of high fidelity, true omnidirectionality provides a greater degree of musical realism and increases the aural perception of the listener. The psycho-acoustic effects of a stereo pair of true Spherical Sound Sources requires many hours of acclimatization, but once the listener's ear has accepted the more complicated aural impulses, the result is the most realistic perception of sound reproduction possible.

It has become a well known fact in recent years, even among audio consumers, that a speaker providing the listener with a combination of direct and reverberant sound imparts a greater sense of spaciousness and realism to the listening room. A truly omnidirectional speaker carries this concept many steps further by providing the maximum possible reverberant field effect, ie., the ratio of reflected energy from an omnidirectional speaker arriving at the ear from all directions, milli-seconds after the direct input, imparts an ambience and realism to the reproduced signal unequalled by any direct or partially reflective speaker.

The amount of reverberant field effect of any speaker is determined by the amount of dispersion, especially of mid and high frequency energy, the amount of reflective versus absorbant surfaces in a listening room, and the speaker's position in relation to those reflective surfaces. In order to effectively utilize the maximum effect of reverberant field in an acceptable listening environment, a speaker must be able to accurately supply the listener's ear with two distinctly different and separately perceived aural inputs: the transient information and the tonal information.

The transient wave form provides the brain with bits of purely digital information. The time of first arrival of the transient at each ear is compared and the difference between these two arrival times provides a directional analysis and the greater part of the stereo image. Without proper transient information the stereo image becomes distorted, possibly through exaggeration of the size of the image, possibly through a minimization of the difference between channels.

The tonal information provides the brain with the actual musical overtones. The ear has the ability to act as a Fourier Analyzer and to separate all the complex tonal input into its actual frequency content. Even the most complicated harmonic structures and overtones are individually analyzed and then transmitted separately to the proper information processing sections of the brain, where the listener enjoyment of the reproduced musical signal occurs.

As long as a speaker provides transient information that can be accurately identified by the ear, the presence of a reverberant field effect serves to multiply and enhance the tonal input from the original musical signal allowing the brain a longer period to identify and enjoy the complexities of the musical overtones. Therefore, the greater the reverberant field, the greater the psycho-acoustic pleasure becomes.

*EPI Inc.

The lack of accurate transient response from the speaker, or an unusual listening environment that would provide an extreme saturation of reflected sounds (the opposite of an anechoic chamber) could result in a muddy or blurred sound. The problem of transient information can be overcome with proper speaker design and an unduly high percentage of reflected sound would never be a problem in a normal listening room.

In a closed acoustical environment such as a normal listening room, an additional benefit of a spherical sound source is its ability to produce a field of sound, in much the same way that the earth produces a gravitational field. Because of the reverberant or reflective characteristics of a room and the psycho-acoustic effects of reverberation, the acoustic field produced is equivalent to a uniform field. In other words, there is no apparent source and no apparent change of loudness in different room positions. Because the stereo effect or image is created by the sound of first arrival, or the transient input, there is no degradation of stereo imaging in these uniform sound fields created by the Spherical Sound Source (transients are radiated equally in all directions). An accurate stereo effect is achieved over the entire listening area, provided only that there is a direct line-of-sight path between each spherical sound source and the listener. The psycho-acoustic effects of reverberation described above enable any normal room to closely approximate the effect of a large concert hall more effectively than any other type of transducer. The uniformity of the acoustic fields enables comfortable listening anywhere in the room, even next to one of the Spherical Sound Sources. Listener fatigue is virtually non-existent.

In order to optimize the acoustic field effects, the room should be divided mentally into equal areas with symmetrical reflective characteristics. The Spherical Sound Sources should be located as close to the center of each such area as possible, in order to maximize the amount of reflection and therefore increase the reverberant field effect occurring in the listening environment. The effects on placement are relatively subtle, however, and truly omnidirectional speakers provide an acceptable level of placement flexibility in most listening rooms.

Achieving a Spherical Sound Source

Conventional bookshelf speakers, whether two-way or three-way, which have the speakers mounted on the same surface in the front of a rectangular box, all have similar directional characteristics. Over some portion of the frequency range of a woofer or tweeter, the drivers operate in a linear fashion whose sound radiation characteristic pattern is hemispherical; that is, the sound radiated off axis as far as 90 degrees in any direction, is equal to the sound radiated straight ahead. (It is a common misconception that speaker's radiate sound mostly in the direction they are facing.)

At very low frequencies, the dispersion pattern is even wider; sound is actually radiated backwards from the speaker —this is why speaker placement affects bass response, there being conspicuously more reflected bass when the speaker is in a corner than when it is in the middle of the room or up in the air. At high frequencies, however, all tweeters become directional, that is, the hemispherical radiation pattern narrows to a straight beam whose diameter is the same as that of the tweeter. In a two inch tweeter this transition occurs between 6 kHz and 8 kHz; in a one inch tweeter it occurs above 13 kHz. In general, good dispersion is achieved only if the diameter of the cone is smaller that the wave length of sound concerned.

By taking four conventional bookshelf speakers with acceptable frequency response, linearity, distortion specifications, and hemispherical dispersion over the entire audio bandwidth, and mounting them with a small enough horizontal separation in a four sided cabinet, it is possible to create a speaker that would closely approximate a Spherical Sound Source.

The cost of building a speaker that meets these criteria is high, due to the complexity of the drivers that must be used. I firmly believe however, that the most devoted audiophiles and music lovers would find the enjoyment received from the complex aural and psycho-acoustic effects described above well worth the expense. Until a physicist or acoustic engineer manages to actually build a working model of the mythical "Pulsating Sphere" the hi fi industry and the consumer must settle for the "Spherical Sound Source" available. Anyone who makes the effort to find and listen to a "Spherical Sound Source" long enough to appreciate its benefits will not be disappointed. Æ

Omnidirectional Radiation

G Sioles

Science and technology as well as the arts have been characterized by controversy, and audio is no exception with such arguments as "pentodes vs. triodes" in amplifiers, the relative importance of measurements vs. listening tests in evaluating loudspeakers being typical. Sometimes the controversy is more imagined than real and derives its substance from insufficient knowledge, or over-simplification. It is the purpose of this article to discuss a recent "controversy" over the relative merits of omni-directional and "conventional" speakers.

Ideally, the performance specification sheet for a loudspeaker should look the same as one for an amplifier, with the exception of a few physical descriptors of one that do not have an easily definable counterpart in the other (e.g. output impedance). A loudspeaker, however, propagates sound in a three-dimensional continuum, whereas the signals processed by the amplifier are propagated in one, a pair of wires. Because of this, an additional important set of data is needed to show how the acoustic power is radiated in the various directions. It is a statement which is not generally discussed in any great detail because representation of the data is cumbersome (imagine looking at sixteen frequency response curves depicting the performance as it varies with direction from the source). But, we would like to discuss this difference between the loudspeaker and other elements in the audio reproduction chain because it is basic.

It is possible to argue, because of the variety of available room placements, that a loudspeaker should radiate uniformly over a solid angle of between π and 4π steradians. Further, the *power* output in free space (not simply axial pressure vs. frequency response) may have a special form to account for the increase in output at low frequencies resulting from wall reflections. It is not acceptable to have a uniform radiation pattern over 4π steradians at low frequencies, becoming directional at middle and high frequencies in such a manner that the net result is a non-uniform pressure vs. frequency characteristic in the reverberant field of the listening room. And yet, this is not uncommon.

Since non-directional behavior or controlled broad directivity is nominally desirable, from where derives the prejudice in some quarters against omnidirectional speakers? First, some speakers considered to be omnidirectional are not, but instead are directed-reflected type radiators. Second, those who feel that omni's are deficient in certain areas may be making generalizations from a very few poor examples. We are not aware, prior to now, of the existence of true omnidirectional speakers as serious contenders in the high performance speaker race. It would seem that omni's are put down in absentia—despite the fact that designers of conventional speakers generally strive to make speakers non-directional over as much of the frequency range as they can manage.

The question more properly may be, are there any true omni speakers? The answer is no. It is exceedingly difficult to produce a speaker that has uniform radiation over a spherical surface in the near field. What happens in the reverberant field (where people normally listen) is another matter. It is possible to produce a speaker which is essentially a true omnidirectional source, as heard in the reverberant field. It does not suffice, however, to place a number of driver units of individually indifferent frequency responses on the surface of a sphere and hope to get good results. True, omni behavior will result but at some cost in frequency response. Suppose we assume a good design—are there problems uniquely associated with omni's, and are they inherent? I do not think so, but a discussion of potential difficulties is worthwhile.

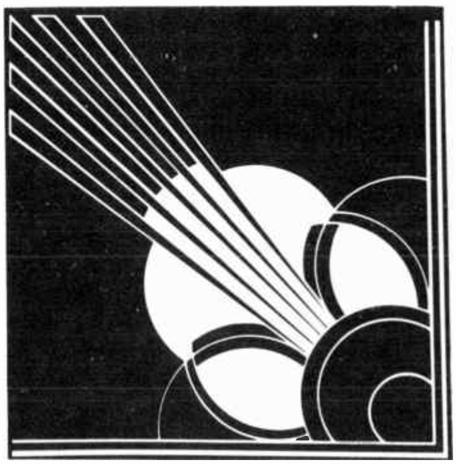
A true omnidirectional source must be either a point source (not possible) or a finite pulsating sphere (not practicable). In practice, an omnidirectional speaker comprises sources so small as to be non-directional as a consequence of their smallness, or sources of known directivity occupying a fraction of a "spherical" surface and equalized so that they radiate constant power vs. frequency, or some combination of the two. If there is any faulting of this approach it may be in the requirement for a multiplicity of sources. What happens is this:

In the frequency range where a number of sources are radiating, the pressure vs. frequency response characteristic will be a function of the microphone position and, in general, will not be "flat". But this is not what we hear. We hear the integrated power output as modified by the listening room characteristics. This poses no problem, if the integrated power output is constant with frequency. There is a possible unlooked-for effect, however, with regard to stereophonic localization. If two multiple driver speakers are so placed with respect to the listener that he does not receive the same "free field" response from both, the stereo images will be imprecise. This may appear to be a significant flaw until one thinks more about the whole process of localization.

Obviously, the problem is potentially most severe if the entire range is covered by a number of drivers, since then the non-uniform response with direction will extend to relatively low frequencies and have more of an effect on the stereo information received by the listener, *if* the speakers are not symmetrically positioned. (If only part of the spectrum is covered by multiple units, it is only the stereo information in this range that may be affected). But, this can be prevented by symmetrical speaker placement. Indeed, symmetry of the listener himself with respect to the two sources is essential to preserve the accuracy of the stereo images, since the process of stereo localization depends on the perception of time and intensity differences between the two channels. These intensity differences are in large measure vitiated by the movement off the axis of symmetry by the listener of approximately one foot. This is because a time of arrival difference of approximately 1 msec. makes necessary an increase of almost 10 db for the later source to be perceived as existing—lacking in this, the sound will appear to come entirely from the near source. Such constraint on the listener is more restrictive than the requirement of symmetrical orientation of speakers. In fact, with omni-directional speakers the tendency to lose the stereo effect is less when the listener moves away from the axis of symmetry—a significant advantage.

Finally, the acoustic characteristics of the listening room are far more important than most people realize. Because the ratio of reverberant to direct sound from omnidirectional speakers is higher than that from more directional types, the effect of the room is correspondingly greater. Since many listening environments (e.g. some audio dealers' showrooms) are less than good acoustically, an omni speaker may come off second best in an A-B listening test with a more directional type. However, for one who does not wish to be fixed in space for his listening enjoyment, and can provide a reasonably good acoustic environment, an omni-directional speaker system is definitely advantageous. Æ

Quadraphony Needs Directional Loud-speakers



***Benjamin B. Bauer**

When stereo began its spectacular rise in popularity more than a dozen years ago I presented a paper before the Audio Engineering Society demonstrating that Stereophonic Perspective could be improved significantly, regardless of listener position by proper design of the directional characteristics of loudspeakers and their placement with respect to the listening area. The improvement is explainable in terms of semi-directional polar patterns of conventional (e.g. "bookshelf-type") loudspeakers; it can further be enhanced with more strongly directional (e.g. "dipole" or "gradient") radiators[2]. A similar analytical and experimental process leads us to conclude that quadraphony also benefits from properly positioned semi-directional and directional sound sources.

What about the role of "omni-directional" loudspeakers we hear so much about? It turns out that true omnidirectional radiation at all frequencies is difficult to achieve in practice; "semi-omnidirectional" performance, however, can be attained with relative ease. Omnidirectional loudspeakers obviously do not require directional orientation to cover a quadraphonic listening area reasonably well and, therefore, often are able to provide quadraphonic performance superior to that obtained with improperly oriented semi-directional loudspeakers; this is a very commendable attribute of systems intended for use by the lay public. On the other hand, omnidirectional loudspeakers can result in nasty wall reflection problems, and furthermore, any knowledgeable Hi-Fi enthusiast, or one who takes a bit of trouble to optimize loudspeaker placement, is apt to gain more satisfaction and improved quadraphonic performance with well designed semi-directional loudspeakers; or, if he is fortunate enough to obtain them or skillful enough to devise them—with properly designed dipole units.

To provide a better understanding of the principles involved in applying directionality to quadraphonic loudspeaker arrays we discuss first the physics of omnidirectional or semi-omnidirectional, semi-directional, and directional loudspeakers.

Omnidirectional Loudspeakers

At first blush, it would appear easy to design an omnidirectional loudspeaker; actually the task is rather formidable. A truly omni-directional radiator is defined by a spherical surface which expands and contracts radially equally and inphase. There are not many practical ways in which such a transducer can be fabricated. One approach is to use a hollow sphere (or two abutting hemispheres) of suitably polarized piezoelectric material (e.g. polycristalline lead zirconium titanate ceramic), including suitable internal and external electrodes to receive the electrical signals. Such a ball vibrates uniformly radiating equal amounts of sound intensity in all directions. Unfortunately such a transducer does not perform efficiently in air, (albeit it works fine underwater).

To improve efficiency we can place a large number of small moving-coil loudspeakers on the surface of a sphere; but since the radiation characteristics of all the units must properly overlap, this approach turns out to be quite complex and expensive. Another possibility is to install a ring of small loudspeakers around a cylindrical drum, or even to place a single transducer on the end of the drum and to confront it with a reflector adapted to direct the medium and high-frequency sounds equally all around into the horizontal plane. Thus, from the truly omni-directional ceramic ball loudspeaker we progress in steps to various practical "semi-omnidirectional" designs which radiate sound relatively uniformly only in the horizontal plane.

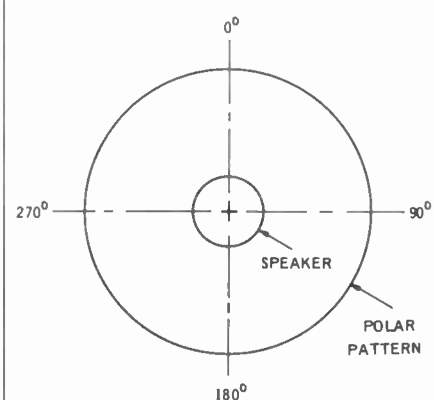


Fig. 1—Polar pattern of an omni-directional speaker in free space.

Graphically we show this uniform radiation in Fig. 1 by a circular "polar pattern," which signifies that the sound pressure radiated at a given distance in a 360° compass is constant.

Next, we consider briefly what happens when an omnidirectional loudspeaker is placed near a reflecting wall or corner. In this circumstance its performance can best be analyzed by the method of virtual images. For example, in Fig. 2(a) the loudspeaker center is placed at 1 ft. from a reflecting wall. Because sound travels at a speed of 34,400 cm/sec, the wavelength at a low frequency, say 50 Hz, is 34,400/50 = 688 cm (22.6 ft.)—much greater than

1. "Broadening the Area of Stereophonic Perception" *Tenth Annual Convention of the A.E.S.*, N.Y., Sept. 29, 1958.
2. B.B.Bauer, *Jour. A.E.S.*, 8, 2, 91-94 (April 1960).

*CBS Laboratories, Stamford, Conn.

the distance between the loudspeaker and its image (2 ft.). Both the real and the virtual source may be assumed to radiate inphase resulting in the doubling of sound pressure, with the polar radiation characteristic remaining nearly circular, as shown by the pattern

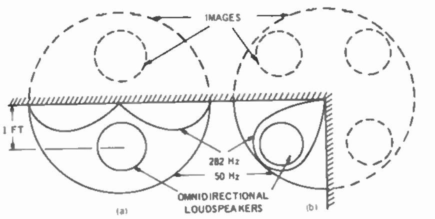


Fig. 2—Polar pattern of omnidirectional loudspeakers placed (a) near a wall (b) near a corner.

labeled "50 Hz" in Fig. 2(a). In reality, real radiation is only outside the wall P shown in solid line. The virtual source and radiation being shown in dash-line. (The presence of this virtual source accounts for increased bass when a loudspeaker is placed near the wall). At higher frequencies, the wavelength λ becomes comparable with the distance between the loudspeaker and its image, resulting in interference patterns. For example, at 282 Hz the distance is precisely ½ λ resulting in total cancellation of radiation in a direction perpendicular to the wall, as shown by the polar pattern labeled "282 Hz." At other frequencies, different patterns will be generated. Placed in a corner, again at 1 ft. from both walls, as shown in Fig. 2(b), three virtual images are formed. Again, at 50 Hz the pattern is nearly circular, and the radiated sound pressure is increased four-fold. At 282 Hz, the radiation near the walls drops to zero, but the radiation along the diagonal is a maximum, resulting in a rather narrow polar radiation pattern. Again, at other frequencies different patterns will be formed.

Therefore, the response from an omnidirectional loudspeaker is rather unpredictable near reflecting walls or corners, suggesting that the presence of acoustical absorption on or near the walls may be desired to avoid the higher-frequency reflection modes.

Semi-Directional Loudspeakers

The majority of "bookshelf-type" and similar loudspeakers are semidirectional. This is to say, they are omnidirectional at low frequencies becoming relatively directional at high frequencies. This is illustrated by the way of example in Fig. 3 where at (a) is portrayed a loudspeaker consisting of a sealed box (popularly known as "infinite baffle") say 12 × 20 in. in cross-section, enclosing a driver with a piston width W = 8" or approximately 20 cm. In actual practice the piston is circular or elliptical; but to make our example as simple as possible we assume it to be rectangular with the long dimension perpendicular to the paper.

At a low frequency, say 50 Hz, where the wavelength is much greater than the dimensions of the box, the particles of air displaced by motions of the piston move to-and-fro together in imaginary channels—much like water flowing from an opening, as illustrated by the streamlines in Fig. 3(a). At a distance from the box, it becomes possible to strike a circular surface along which all the streamlines are distributed with a near equal density of flow, corresponding to equal sound pressure which expands in concentric circles away from the loudspeaker. Under this circumstance the loudspeaker behaves like an omnidirectional radiator.

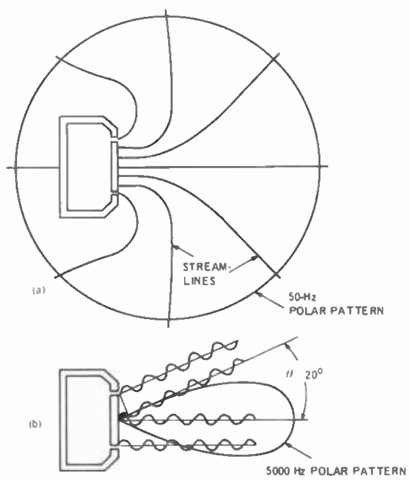


Fig. 3—Polar patterns for a piston in closed box, for (a) low frequency (50 Hz) and (b) moderately high frequency (5000 Hz)

As the frequency increases the wavelength becomes progressively shorter with the consequence that sounds from various portions of the piston are no longer inphase causing the radiation off the principal axis to be diminished or even to become completely cancelled. For example, consider the situation at 5000 Hz where λ is but 34,400/5000 = 6.88 cm (2.7 in.). As may be seen in Fig. 3 (b), the wavelets from the center and edge of the piston in the direction parallel with the axis are in additive phase resulting in intense sound radiation forward of the piston; but at some angle, θ, the wavelets are found to be in phase-opposition causing complete cancellation. In the example given, θ is readily found as follows:

$$\sin\theta = (\lambda/2)/(W/2) = \lambda/W \quad (1)$$

or, since $\lambda = 6.88$ cm, and $W = 20$ cm, $\sin\theta = 6.88/20 = 0.344$; or $\theta = 20°$. From its on-axis maximum, the radiated sound pressure progressively diminishes to zero as shown by the polar-pattern in heavy line in 2(b). It is obvious that, in this last example, we are dealing with a narrow directional pattern which is unsuitable for high-quality sound reproduction at widely spaced positions in the room. To "broaden" the directional pattern sufficiently to obtain reasonably good coverage, we must restrict the upper frequency at which a piston is allowed to radiate. A workable rule of thumb for *circular* pistons is that the wavelength should not be less than the diameter of the piston. For an 8-inch piston this corresponds to 1720 Hz. To provide a satisfactory radiation pattern to 20,000 Hz the diameter of the piston should no exceed approximately 34,400/20,000 = 1.7 cm (0.68 in.).

At this point the reader might wonder why not use the small piston for all frequencies simply by making it work that much harder at low frequencies. This approach is counter-productive because the sound pressure generated at any given frequency at any point in space is related to the volume of air displaced by the motion of the piston, i.e. by its area multiplied by its linear vibration amplitude. Thus, an 8-inch diameter circular piston vibrating with a ¼-inch motion is apt to provide adequate bass sound; a 1.7-cm diameter piston which has 1/22 the area of an 8-inch piston would have to have a 22-times longer stroke for the same sound output. —or 5½ in. which obviously is impractical.

Thus, the designer is caught between the limitations of maximum allowable piston amplitude, at one end of the frequency scale, with the directional radiation problems at the other, and he has to allocate the range covered by each piston to a relatively limited band of frequencies. This explains why a superior loudspeaker system usually will employ several drivers of different diameters interconnected electrically with dividing networks to convey to each its proper portion of the spectrum. Typically, the response pattern of a good semi-directional loudspeaker is omnidirectional (circular) at low frequency narrowing down to a 90°-60° included angle at about 1000 Hz and remaining within this range up to the highest frequency of interest, as various radiators of progressively smaller size

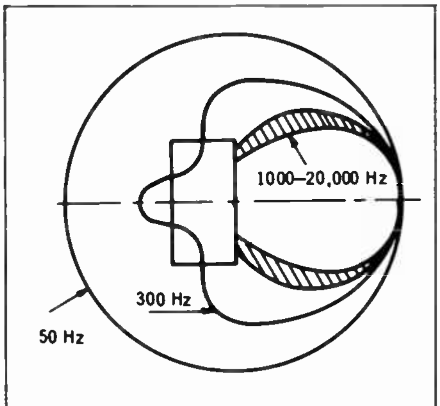


Fig. 4—Typical polar patterns for a conventional semi-directional (e.g. bookshelf-type) loudspeaker at various frequencies.

come into play, as shown in Fig. 4. Because the baffle box diminishes the radiation toward the back at mid-frequencies, such a semidirectional loudspeaker is less bothered by reflections from the walls or corners, than an omnidirectional unit. Without pretense of offering a treatise on loudspeaker design, it should be noted that the polar pattern of a loudspeaker cone can be modified by shaping, adjusting compliance, adding acoustical lenses etc.

Dipole Loudspeakers

A dipole loudspeaker is simply a piston (e.g. loudspeaker mechanism) vibrating in open air without an enclosure to confine the back radiation. The efficiency of such a device is a function of frequency because the radiations emerging from each side of the piston tend progressively to cancel each other as the wavelength increases. Efficiency may be improved by adding to the piston a baffle, as shown in Fig. 5, which increases the distance between the front and the back of the piston. It is easy from Fig. 5 to visualize that, in the perpendicular front, or zero degree direction, the back sound radiation has to travel an added distance D before it can proceed to the front; the added distance giving rise to a phase differential between the two waves producing a net sound pressure at a given point in space designated as P_0. As we move in a circle to the 90° direction, the radiations from both sides become equal and in antiphase; thus there is a zone of silence at all points on a surface S perpendicular to the axis. As one travels to the back, or 180° direction, the distance D comes again into play (except that this time, from front-to-back) resulting in a maximum sound pressure at P_{180}. It is not difficult to prove mathematically that as one moves around the circle, the pressure function follows a cosine law, the polar pattern taking on the form of two circles at both sides of the piston. To avoid an excessive loss of efficiency the dimension of the baffle should be no less than approximately $\frac{1}{6}$th the wavelength of the lowest frequency of interest. At high frequency the radiation from the piston narrows down in a manner similar to that described in Fig. 3. Therefore, the highest operating frequency for any one piston should be that corresponding to a wavelength equal to its dimensions. An 8-inch diameter loudspeaker installed in a 13-14-inch baffle, has a satisfactory operating range between 250 and 1700 Hz. A second, correspondingly smaller gradient loudspeaker would be needed to cover a range between 1700 Hz and 10,000 Hz, etc. The polar pattern obtained with a composite gradient loudspeaker would then be approximated by the broadened circular outlines in Fig. 5.

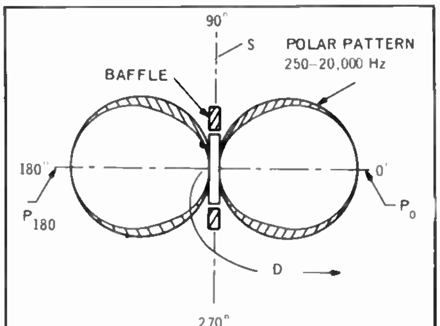


Fig. 5—Theoretical polar pattern of a dipole loudspeaker composed of progressively smaller dipole elements to cover the full frequency range between 250-20,000 Hz. (Below 250 Hz, an omni-directional loudspeaker is used).

Experience has shown that response below 250 Hz can readily be provided with a conventional (omnidirectional) loudspeaker in an infinite baffle, without greatly influencing the aurally perceived directional characteristic of the gradient array.

Thus, a practical dipole loudspeaker generally is omnidirectional below about 250 Hz, and exhibits a figure-eight pattern above 250 Hz, as shown by the heavy outline in Fig. 5. The frequency response of the combination must be carefully tailored to be "flat" overall. The biggest advantage offered by the dipole loudspeaker array is that it retains its cosine-law directional pattern over that portion of frequency which conveys the major part of directional information i.e. between 250-20,000 Hz.

A few dipole loudspeakers are available commercially. Usually they employ electrostatic high frequency sections and a moving coil bass section. For the present, however, the majority of high fidelity enthusiasts will have to be content with semi-directional loudspeakers to obtain the improved area coverage described below.

Application of Directional Loudspeaker to Stereophonic Arrays

The wrong and the right way of placing conventional semi-directional loudspeakers (e.g. bookshelf loudspeakers) for stereophonic listening is shown in Figs. 6 and 7, reproduced here from the paper referred to previously[2] with the kind permission of the Audio Engineering Society. In the first case, the loudspeakers are placed parallel to the front wall of the room. A listener at position P is at a greater distance from loudspeaker A than from the loudspeaker B. Also, the radiation strength of loudspeaker A in the direction P, represented by the vector AQ, is smaller than the radiation strength of loudspeaker B in the direction BP, as portrayed by the longer vector BR. Thus, the loudspeaker B has two advantages: The listener hears predominantly the sound of B and very little or no sound from the loudspeaker A. Furthermore, any sounds panned in between the two channels appear to move strongly towards B, because the Haas or "precedence" effect tends to credit the nearest loudspeaker with being the source of sound.

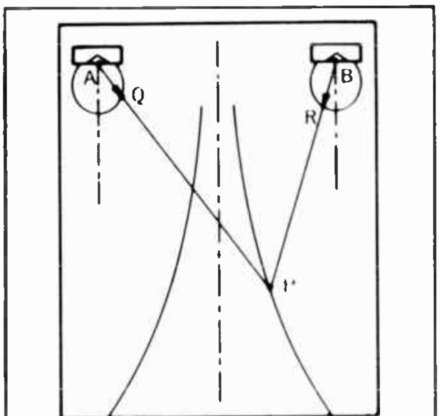


Fig. 6—Relatively narrow stereophonic listening area is obtained with directional loudspeakers placed parallel to the wall (*Courtesy AES.*)

Next, we examine the operation of the improved placement method in Fig. 7. Here the listener at point P again is at a greater distance from A than from B. However, because the loudspeakers are at an appropriate angle with respect to each other and with respect to the listener at P, the radiation vector AQ in the direction AP is greater than the radiation vector BR in the direction BP. Thus, the distance effect is, in part, compensated by the directional effect. The sounds arriving from the two loudspeakers remain in balance over a considerably broader area than is possible with the arrangement in Fig. 6. Also, the added

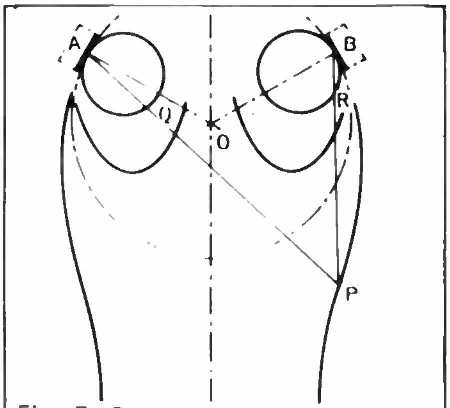


Fig. 7—Broadened listening area obtained with directional loudspeakers angled towards the center (*Courtesy AES.*)

signal strength from the farthest loudspeaker tends, in part, to compensate for the Haas effect thereby allowing a significant amount of common sounds to appear in between the loudspeakers avoiding the so-called "hole in the middle" effect.

The solid line contours in Figs. 6 and 7 represent the listening area within which the signal strengths from loudspeakers having radiation pattern as described remain within 3 dB of balance relative to each other. The advantage of the inclined orientation is evident. For an average bookshelf-type loudspeaker, the included angle between the loudspeaker axes, for proper orientation, turns out to be approximately 120-130 degrees.

Wall Reflections with Dipole Loudspeakers

We have shown, in a previous section, that omnidirectional loudspeakers under certain circumstances are significantly affected by reflections from the walls. Dipole loudspeakers, by contrast, are relatively free of this problem.

It has been seen from one of the preceding sections that directional loudspeakers normally are placed at an angle to the walls enclosing the listening area. Fig. 8(a) represents the polar pattern of a dipole loudspeaker placed, say, at 45° to a wall. It will be noted that a listener at P is subjected to its maximum output. This same listener is in a near-null orientation with respect to the virtual image caused by wall reflection. Furthermore, with the dipole loudspeaker placed in a corner, as shown in Fig. 8(b), two of three virtual images are oriented with the null planes toward the listener, thus contributing relatively little to the sound pressure at P. Thus, the response characteristic produced by a dipole loudspeaker is apt to be less affected by room acoustics than that produced by an omnidirectional unit. It should be noted, however, that the "back" radiation of the gradient loudspeaker does exist, and while it may not be significant on first reflection, it may become so for subsequent, later, reflections helping to create a desirable "ambiance" effect.

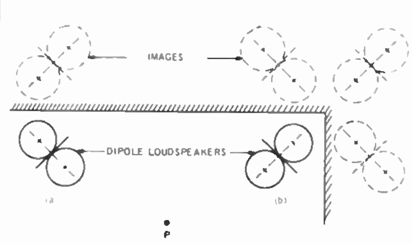


Fig. 8—Angled dipole units (a) near wall and (b) near corner.

Quadraphonic Arrays

Following the example given previously in connection with stereophonic arrays, the benefits of directional loudspeakers in quadraphonic listening will now be demonstrated. Fig. 9 portrays a quadraphonic listening area with 4 omnidirectional loudspeakers placed in the corners. For the sake of simplicity we assume for the moment that there are no walls to cause directional cancellation and reinforcement problems. Thus, the polar patterns of the loudspeakers are shown as four circles concentric with the corners of the listening area (which forms a dash-line perimeter). That portion of the listening area where the sound pressure from any of the four loudspeakers varies no more than ± 3 dB is shown by the diamond-shaped outline.

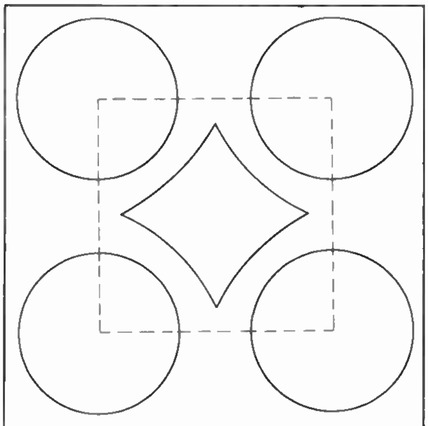


Fig. 9—Quadraphonic listening area with four omnidirectional loudspeakers placed in the corners.

In Fig. 10, dipole loudspeakers are used. The ±3 dB contour is now increased substantially, reaching all the way to the edge of the square; thus

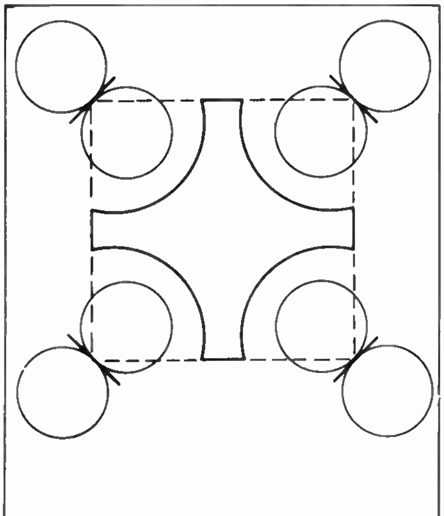


Fig. 10—Quadraphonic listening area with four dipole loudspeakers.

greatly increasing the positional freedom of the listeners. Similar, but perhaps not quite as dramatic improvement is obtained with semi-directional loudspeakers in the corners facing the center of the area. On the other hand, if these semi-directional loudspeakers were to be placed squarely against a wall (instead of being angled), their directional patterns would tend to augment the effect of the inverse distance law causing the optimum listening area further to be restricted.

The aforementioned analysis holds true even in conventional semi-reverberant listening rooms because directional localization depends principally on the sounds of first arrival.

Conclusion

Much remains to be learned about optimum design and placement of quadraphonic loudspeaker arrays. Omnidirectional loudspeakers in vogue today are a partial answer to this problem —albeit one troubled by excessive dependence upon the acoustical characteristics of the room boundaries. Also, omnidirectional loudspeakers produce a relatively limited area of quadraphonic perception. Directional loudspeakers—e.g. semi-directional or "bookshelf" types, and especially the dipole types, are apt to result in a more balanced sound field over a broader listening area, but usually require some experimentation with orientation and seating arrangements. Æ

In All Directions

*John Crabbe

A proper understanding of the mechanism of stereo perception requires extended reading, in general terms or at greater depth for those with a mathematical bent[1], but for my present purpose it can be safely stated that the accuracy with which stereo images may be localised by the listener depends on four factors:

(i) Clearly differentiated electrical information in the stereo signal, given either by Blumlein-inspired coincident microphone techniques, or by unambiguous amplitude pan-potting of discrete signals on to the soundstage.

(ii) Use of identical or near-identical loudspeakers.

(iii) An unobstructed sound path between each loudspeaker and the listener's ears.

(iv) Either an equal path length from the listener's head to both loudspeakers *or*, if the listener is placed to one side, a radiation pattern from the speakers which compensates subjectively for the resulting time differential.

We must assume that the first condition is satisfied, which is reasonable at least for the direct instrumental sounds in most modern recordings (almost invariably pan-potted), though not for the reverberation, which tends to be anomalous. Most good loudspeakers should satisfy the second point except for laboratory measurements, and the third requirement is a matter of common-sense usage.

With perfect two-channel stereo reproduction the full panoply of sound-sources is heard accurately displayed between and beyond the loudspeakers. This accuracy applies not only to the direction of individual instruments or voices, but also to their apparent widths. Now it so happens that in any system that is well balanced and has adequate electrical separation between signal paths, the performance with a central (double-mono) signal is a reliable indication of overall stereo accuracy. If a left-only signal produces a narrow sound image from the left-hand speaker, a right-only signal likewise from the RH speaker, and a double-mono signal produces a narrow image centrally placed between the speakers, then it follows automatically that a stereo signal will be reproduced accurately right across the sound-stage. Unfortunately, this perfect stereo can normally only be obtained if the listener is equidistant from both speakers —that is, if he is in the 'stereo set' placed on the apex of an isosceles triangle subtended by the speakers.

Any reader who doubts this can try it for himself: it will be found that there is a precise listening line along which a double-mono signal is heard as a very narrow and clearly detached sound-source. If this elementary first-step fails there is something faulty somewhere—either in the speakers, the room or the ears! Movement to one side from this ideal position normally causes two things to happen: (i) the image shifts more or less with the listener, and (ii) it broadens and is therefore less precisely located. In any system, insofar as mono does *not* sound as if it were coming from a separate central speaker, there is some falsification of stereo signals—an element of pseudo-stereophony. A major problem in domestic sound reproduction is to minimise this effect over a reasonable listening area, thus providing good stereo for practical use in the home.

The normal approach to this stabilisation of stereo images is to employ to best advantage any directional characteristics possessed by the speakers. Essentially, the central image (and everything else with it) becomes distorted when listening away from the bisecting line because one is then nearer to one speaker than the other, which gives its signals a time-lead. Because of Haas-effect (precedence-effect), this lead in time produces a subjective boost in loudness from that channel, which shifts and broadens the apparent sound-source in that direction. Now, if movement to one side resulted in a *lower* acoustic level at the listener's ears from the speaker on that side (and/or a higher level from the other side), the image-shift due to a time differential could be cancelled by a contrary shift due to the loudness change. This is the basis of the well-known Hugh Brittain loudspeaker placing[2], pursued more fully to overcome certain anomalies by Joseph Enock.

Practical loudspeakers vary enormously in the shape and frequency-dependence of their forward radiation patterns, and since an ideal 'Enock' speaker would have one particular lobe shape and no tendency at all to extra beaming at high frequencies, it is evident that the whole business is full of compromise. However, with patience most conventional speakers can be made to perform quite satisfactorily in most rooms. Setting up may be a tedious business[3], and it may sometimes involve very curious angling as advocated from time to time by Ralph West in his speaker reviews. But my experience is that if one is prepared to sit fairly well back from the speakers and not unreasonably out to the left or right extremities, it is possible to obtain good stereo over a sensible listening area. By 'good stereo' I don't mean the pin-pointed accuracy heard from the stereo seat, but a fairly consistent and well defined sound-stage of the sort associated with a double-mono signal that never shifts more than a third of the way towards one side or broadens to an angular width of more than about five degrees.

What has all this to do with omni-directional speakers or their advertising? Taking the second point first, it is extremely relevant, for we have been shown families of seven people ridiculously huddled around one chair in the middle of a room whose only other contents are a pair of conventional speakers, an amplifier and a player. This is a gross falsification of the domestic listening situation, attempting to create a myth that until recently it has been necessary to upset one's living arrangements in this manner in order to enjoy the benefits of stereophony. Even a hi-fi dealer wrote to me in support of this extremist position, conceding that 'there is a place for the lone listener in his throne the stereo seat' who can 'choose from a mass of direct sound speakers . . . but there are many more readers with a family and friends who like to sit round the fire-side', etc. Now it is true that sitting in a semi-circle around a fire does create difficulties for desiderata (iii) and (iv) listed earlier, but I suggest that this is only one special case among endless domestic possibilities, and that it is unfair to adopt such an extreme 'either—or' attitude about those who listen to music in their homes.

1. *Stereophony* by N. V. Franssen. (Philips Technical Library).
2. *Two-channel Stereophonic Sound Systems* by F. H. Brittain and D. M. Leakey. 'Wireless World', May/July 1956.
3. *Installation: Loudspeakers* pp. 223-229. 'Hi-Fi in the Home' by John Crabbe (Blandford Press).
4. *Two Channel Quadraphony* by David Hafler. 'Hi-Fi News', August 1970. (See also 'A New Quadraphonic System' David Hafler, 'Audio' July 1970.)

Editor**, ***British Hi Fi News**.**

(abridged version

In any case, stereo is really a fairly subtle business and can only be appreciated fully by those who *listen* to music—it is hardly necessary for background while sitting around the fire!

The other important point about the adverts is their claim that omnis surround the listener with stereo sound wherever he or she is in the room, obviating the supposed need for 'stereo seat' listening and implying that the type of stereo obtained on the bisecting line with conventional speakers is achieved everywhere with omnis. This is where my earlier remarks about stereo perception come into the argument, for it can be shown both theoretically and practically that omni-directional loudspeakers distort the stereo sound picture more or less severely.

Firstly, they cannot by definition offer a sound intensity pattern that compensates for precedence-effect because they radiate equally in all directions; thus even in an anechoic room there would be considerable shifting and broadening of a centre-stage image as heard by an off-centre listener. Secondly, in a normal room there is relatively little direct sound from omnis of the Sonab type (without *any* forward radiating unit) so that the ears are presented with a very complex series of confusing reflected wavefronts which upset the localising faculty. This means that even in the stereo seat a nominally central sound-source seems vague and broad in most rooms, the only really precise directional information (if the room permits any at all) arising from extreme left or right sounds. In my own sitting room, which is acoustically rather 'dead' compared with most and therefore relatively disinclined to scatter the stereo sound-picture, a pair of Sonab OA-5s was quite incapable of producing anything remotely approaching a narrow sound-source from a double-mono signal. On a stereo recording of a harpsichord concerto on which a seemingly small solo instrument is contrasted nicely with a broad orchestral backcloth, the harpsichord stubbornly occupied the full space between the speakers as heard from any point in the normal listening area.

This is not good stereo—it is hardly stereo at all—and I must beg to differ most strongly with critics who state that omnis 'do provide a good stereo image virtually anywhere in the room'. They do not and they cannot. Neither can they provide a satisfactory and reasonably consistent frequency response from sample to sample, depending as they do entirely on the environment in which they are used; this is contrary to all good loudspeaker design criteria. Despite all this there are bound to be a few freak rooms in which it is impossible to obtain a satisfactory listening area with conventional speakers but which reflect the sound from omnis in a manner that happens to provide some compensation for Haas-effect in a pseudo-Brittain fashion. Any readers with such rooms (one was amongst my correspondents) may ignore the bulk of this article—but my general thesis stands.

Some people not in this special category may nevertheless *like* the sounds produced and many will welcome the fairly constant type of sound pattern throughout the listening room that was mentioned and praised by Donald Aldous in his review of the Sonab in November. Some have referred to this review as if it vindicated their viewpoint, apparently failing to notice that Donald did not claim that the relatively stable sound-field represented good stereophony. Indeed, he scattered a fair number of serious doubts, stating that 'there is loss of definition and precise images', that it is 'true that stereo is often anomalous . . . and this may prove disconcerting, especially to the more experienced listener', and that 'it is essential that the reader should be aware that the contention concerning directionality, at least, is fallacious when related to sound *reproduction*'. It is a case of distortion that remains equally distorted from all points of view!

I think that covers the objective side of the matter and explains why we commented so adversely on the Sonab advertising—though I see that more recently we have been asked to believe that these loudspeakers have some curious extra property enabling them to reproduce the quarter tone scale of Indian music that is 'too much for most systems'. It's certainly too much for me—I give up!

On the musical and subjective side there is much more room for argument and maneuvre. Once the supporters of omni-directional speakers have admitted that they generally lose a lot of directional information and suffer from rather extreme distortions of lateral perspective, then I will admit that they may indeed actually prefer this sort of sound and that they have every right to. But it must be understood that in terms of sound *reproduction*, of producing an accurate acoustic equivalent of the signals passing through the stereo amplifier, omni-directional loudspeakers represent a firm step backwards. Musically, this may not seem to be the case but if so this can only be due to other limitations of two-channel stereo which are receiving partial compensation via the loudspeakers. This indeed is part of the Sonab philosophy, emanating from Stig Carlsson, the argument being that in real life most of the sound energy arriving at our ears in the concert hall comes via reflections. This was outlined in the November review and is a point that has been made on many occasions when discussing the philosophy of stereo reproduction. It is basic also to the Bose loudspeaker, though this is in a rather different category to the Sonab, without the latter's flimsy construction and rather obvious colorations, and with at least one forward-facing drive unit. However, developments in quadraphony or pseudo-quadraphony promise a more satisfactory type of solution, taking us much closer to a live concert-hall atmosphere than the rather unreliable use of multiple short room reflections via omni speakers.

I am sure that it is this missing sense of all-round atmosphere that leads people to look beyond conventional stereo, with its sound-stage at one end of the room and no reverberation from around or behind the listener. But things are now on the move, and even limited experiments with 'difference' signals[3,4] can be a revelation in added spaciousness compared with the effects achieved by omni speakers. And there is no penalty to pay in the accuracy of spatial reproduction on the forward sound-stage.

Several of my correspondents were slightly offended by the phrase 'undifferentiated wodge of sound' used to describe the omni type stereo picture. The dealer whom I mentioned earlier pointed out that in his view this is just what many people want and that it gives a great deal of musical pleasure. Well, that may be so for some stereo beginners, especially if their taste is for big, lush orchestral music—Strauss tone-poems for instance—just as upward-facing column speakers were all the rage for a while when stereo reordings were first introduced. We have been through all this great debate before; but gradually, as people listened more carefully and became more critical, they came to realise that what they thought was stereo was really little more than mono thrown around somewhat by two speakers—in fact an undifferentiated wodge.

One reader claimed in a letter that omni speakers are 'as great an improvement over ordinary stereo speakers as stereo itself is over mono'. Well now, if this is so it would follow that to switch a pair of omnis from mono to stereo would be at least as revealing or dramatic as a similar switch using conventional speakers. But it is generally a good deal *less* revealing, for the simple reason that omnis dilute the stereo image and inflate a mono signal to the point where they are rather similar.

Finally, a few words in favour of the

musical subtleties of conventional stereo, subtleties not demanding bisecting-line listening accuracy, but simply ordinary loudspeakers and ordinary seating sensibly arranged in an ordinary room.

This was exactly what happened when I played the aforementioned harpsichord concerto recording: on conventional speakers in mono the whole orchestra and solo instrument appeared to occupy a fairly narrow band in the centre of the speaker wall, while in stereo the orchestra spread out correctly and grandly in its various sections with the harpsichord remaining of slender proportions at front-centre; on omni speakers in mono both harpsichord and orchestra appeared to occupy the whole wall, and in stereo the only change was a suspicion of upper strings more prominent on the left. The moral of this story is that if you want a stereo recording to make an impression on omni speakers you must exaggerate the left/right instrumental separation and minimise centrally placed sources for all you are worth—a thoroughly unmusical and reprehensible business, yet my same correspondent goes on to say that he 'looks forward to the record industry catching up with the equipment manufacturers by producing records suitable for reproduction on these omni-directional speakers'. God forbid! listen to the sound first from the stereo seat and then from a point far enough to one side to shift and stretch the sound image unreasonably. My ears register a change of tonal quality which seems to be independent of HF beaming effects. Tone-colours are part of music, so this sort of thing must affect musical pleasure at some level.

Much music demands, and some conductors use, spatially separated 1st and 2nd violins. Done discreetly, as on many recordings, the two string groups are placed to left and right of stage-centre, but not pulled apart ridiculously. A lot of delightful antiphonal effects are there for the hearing, but they are certainly less easily distinguished in a 'wodge' of sound. Solo instruments set against an orchestral backcloth sound quite unnatural if stretched out in the manner of the harpsichord already mentioned; in violin concertos, particularly, some of the musical drama is dissipated if the instrument's physical smallness is lost. This applies also to voices, especially in opera where both subtlety of movement or placing, and moments of high drama, may be lost or even contradicted in the proverbial sonic wodge.

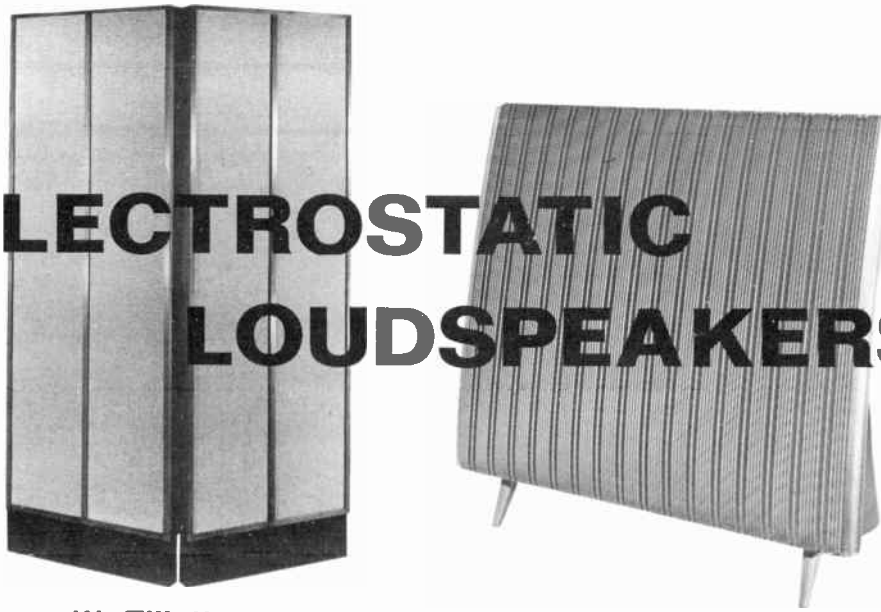
Complex many-stranded counterpoint is sometimes difficult to follow without the aid of a score, especially when the music is for multiple divided strings and therefore unsignposted by a variety of instrumental timbres. Such music benefits from good stereo because of the audible but often subtle separation in space. Finally, chamber music, and particularly the string quartet, which can sound so very convincing when well reproduced but quite vague and silly when distorted by omni loudspeakers. Anyone with experience of listening to a real quartet at fairly close quarters soon realises the absurdity of the freakish quasi-stereo offered even by a moderately differentiated 'wodge'.

This all means that sooner or later people will get fed up with omni-directional loudspeakers—just as most people eventually abandoned their column speaker about ten years ago. (There is a possible analogy here with headphone listening, the present popularity of which—due to its consistently accurate stereo—could be a reaction against the vague stereo heard even from improperly used conventional loudspeakers.) Singers' mouths or solo violins several feet wide which cannot be placed at all certainly in an particular direction are tiring and irritating to live with. They will come home to roost. This I know from personal experience, having been a keen advocate of reflected sound not many years ago! Æ

SUMMING IT UP

Summing up: there is no doubt that omni-directional speakers or systems that specifically use walls for reflection do give a more spacious kind of sound. Under the right circumstances, one is less aware that one is listening to two loudspeakers. It is also true that this effect is achieved at the cost of definition. On the other hand, very directional loudspeakers give a sharp stereo image but the listening area is restricted. In the early days of stereo (two channel) I maintained that the optimum dispersion angle was 120 degrees but in these days of 16-channel mixers and multi-mic techniques I cannot be so dogmatic. Stereo itself is an illusion and the program material goes through many processes of mixing, dubbing, equalising and so on. Some producers exaggerate separation, some transport the listeners to the conductor's podium and others try and give him the impression of being in the middle of the 10th row back. Then again, most of today's music is recorded in the studios—not the concert hall at all! Finally, there is the question of room acoustics. The room must be considered acoustically as an extension of the loudspeakers and what sounds superb in one room can be incredibly bad in another.

Perhaps the best answer to some if not all of these problems lies with the intelligent use of the quadraphonic medium. This can give us a better sound image without relying on random room reflections or being so affected by room acoustics—especially standing waves. Moreover, as Jim Long stated in his recent article on microphones, "Four mic/four channel recording reduces the need for accent microphones. The ability of four-channel stereo to sort out a single event amidst complex aural confusion—if the recording is properly handled—can be downright uncanny!" The big question will be: What kind of loudspeaker radiation pattern will give best results with quadraphonic sound? My own tests indicate a dispersion of 90 degrees but I am reserving judgment for the moment. *G. W. T.*



ELECTROSTATIC LOUDSPEAKERS

George W. Tillett

The principles of electrostatics as applied to transducers have been known for many years and a complete electrostatic telephone system was actually demonstrated by one Dolbear at the Paris Electrical Exhibition as long ago as 1881! Electrostatic loudspeakers did not make their appearance until about 1925 and most of the development work for the next fifteen years or so was done in Germany by such pioneer as Ernst Klar and Hans Voght but American physicists like McLachlan, Kellogg and Hanna were also active in this field. In its simplest form an electrostatic speaker consists of a conductive-coated film diaphragm and a metal plate as shown in Figure 1. If a signal is applied between the two, the diaphragm will vibrate accordingly. However, it will always move *towards* the plate as the voltage can only cause an attraction. Consequently, severe distortion will be produced and if a sine wave is applied the speaker will function as a kind of rectifier i.e. both positive and negative pulses will cause the diaphragm to move towards the plate. If a d.c. bias voltage is applied then the diaphragm will be partially compressed and a signal voltage can then cause it to move *both* ways about its biased position. The space between the diaphragm and plate must be very, very small and if an attempt is made to increase the distance so the speaker can can work at lower frequencies then the efficiency falls drastically. Moreover, the distortion will increase due to the force varying with the diaphragm position. Thus, simple 'single-end' electrostatic speakers can only be used for high frequencies. A typical treble unit-popular in Europe in the late 30's and just after the War would measure about 3 inches square and the film diaphragm would actually rest on the metal plate. In operation, such speakers would be connected directly to the anode of the output tube of the receiver, thus using a d.c. bias of some 200 volts.

Some of the more expensive units used two plates in a push-pull arrangement as shown in Fig. 2. Applied force is less dependent on diaphragm position and distortion will be much lower. Obviously, the plates must be perforated to pass the sound. Full-range electrostatic speakers follow the same principles but there are certain problems. To get down to the very low frequencies the gap between the diaphragm and plates must be large enough to allow the diaphragm to move. As mentioned above, this seriously reduces the efficiency. To some extent this can be made up by increasing the bias voltage but we eventually reach a limit determined by a corona or voltage breakdown. The practical maximum voltage is usually calculated by making the breakdown point equal to the bias voltage plus half the signal voltage. Even so, the diaphragm area has to be quite large to radiate enough power at low frequencies with reasonable efficiency. Another problem that worried early designers was the distortion produced by the variation of the diaphragm charge. (See Fig. 3). If a strong signal pulls the film to one side then the capacitances of the two halves will become unequal and the total capacity will be increased beyond the normal C1 plus C2 figure because the net increase is greater than the decrease. This means that the corresponding charge will be higher and the diaphragm displacement will be greater than required. In other words, it would produce a kind of amplitude distortion progressively worse at low frequencies. Fortunately, the cure is fairly simple and it merely entails supplying the charge in the diaphragm via a high resistance. If the time-constant introduced is appreciable compared with the lowest frequency to be reproduced then there can be no significant current flow to or from the diaphragm and the charge is *constant*. Well almost, because there is yet another snag. Figure 4 shows a diaphragm pulled over to one side by the signal with one part nearer the plate P1. It is impossible to maintain 100 per cent accurate spacing and so the charge tends to migrate to a point nearest a plate. What happens then? The voltage would fall elsewhere and so the effect is cumulative and could eventually lead to ionization and actual voltage breakdown-quite apart from causing distortion (I). Again the cure is relatively simple and the answer is to make the diaphragm itself a high resistance so the charge cannot move about. Colloidal graphite or vacuum formed gold are two materials used and typical resistance is between 50 and 100 megohms per inch.

How about dispersion? A push-pull speaker will have a figure-of-eight or doublet configuration although experimental models have been made with the rear radiation suppresed—or partially so. If the width of a flat sound source is equivalent to more than one wave-

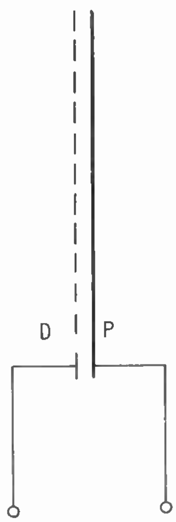


Fig. 1—Showing simple ESL arrangement.

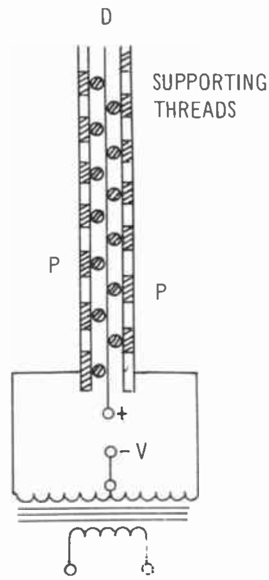


Fig. 2—Push-pull construction. PP are perforated metal plates (sometimes curved) D is the diaphragm.

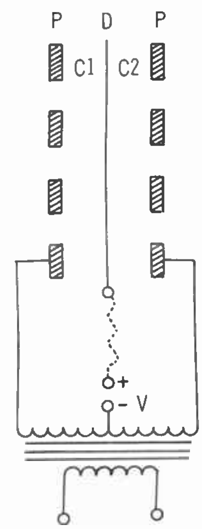


Fig. 3—Push-pull configuration with transformer.

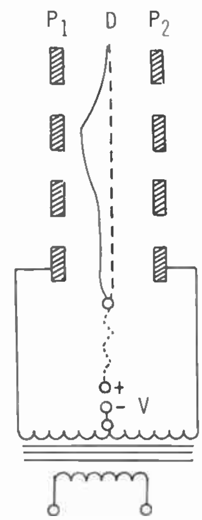


Fig. 4—Showing diaphragm displacement.

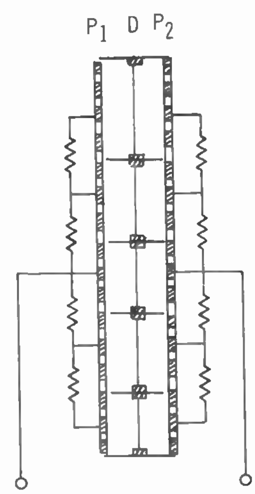


Fig. 5—Method of signal grading or distribution.

length of the lowest radiated frequency, dispersion will be poor with lobes and beaming. One solution is to curve the radiating surface by using several diaphragms spaced in the form of an arc. Another method is to use narrow strips connected as shown in Fig. 5. All the signal is fed to the center pair and resistors progressively reduce the levels at each side (2). A separate bass diaphragm would have to be employed and in the Quad model this is in two sections placed each side of the central treble unit. The step-up transformer is specially designed to also act as a cross-over by the ingenious method of using the leakage inductance in conjunction with the capacity of the actual bass unit. Roll-off is about 12 dB per octave.

It has been said that the diaphragm of an ESL does not 'break-up' as it is driven as an entity. This is not true because the mechanical impedance is not constant due to the edge clamping where it is partly stiffness controlled. It can be shown that nodes and anti-nodes do exist-just as in cone speakers (3).

The inherent distortion of an ESL is very low and Professor Hunt quotes figures of less than 0.5 per cent second harmonic under worst conditions of unbalance (4). Transient response is also extremely good as the moving mass is small. On the other hand, overall efficiency of models at present available is still on the low side-probably not a serious disadvantage-and extreme low frequencies cannot compare with good dynamic systems having 15 or 18-inch dynamic woofers. Because of the doublet configuration, minimum power is radiated at right angles to the axis and so fewer modes of room resonance are excited. In theory, maximum bass radiation will occur when the speaker is placed in the center of a room—in other words the diaphragm impedance is matched better by the lower radiation impedance but in practise the best position is usually two or three feet from a wall. Finally, early ESL's were considerably superior to contemporary dynamic types in respect to coloration. No enclosure—so no box coloration. . . . However, new techniques, better cone materials and enclosure designs have improved conventional systems enormously during the past few years and there are now several dynamic speakers that can stand comparison with the best ESL units. Pros and cons were neatly summed up by Peter Walker, inventor of the Quad speaker some years ago "The horse and the motor car are both effective forms of transport, but the car is not very good at jumping a five-barred gate, nor is it seen at its best in a ploughed field." Æ

1. "Full-range electrostatic speakers". Wireless World, May, June and August, 1955. An alternative design was proposed by H.J. Leak and A.B. Sarkar. This was a reversal of the type shown and the plate is in the middle with a diaphragm each side.
2. Another possibility is the use of diaphragms having a graded resistance. Some models use separate LF and HF transformers which gives some flexibility.
3. "Loudspeakers" by Gilbert Briggs. Cahners Publishing. Boston Mass. See also "Loudspeakers" by E. Jordan. Focul Press, New York.
4. "Electroacoustics" by Professor F. Hunt. Harvard University Press.

The Hows, Wheres, & Whys of Testing High Quality Loudspeakers

Part one **H. D. Harwood***

THE OBJECTIVE TESTING of loudspeakers in a free-field environment has often been attacked on the grounds that they are not listened to under these conditions and that in any case there are subtle effects which are not amenable to measurement.

Whilst these arguments contain a certain amount of truth, there is no reason why we should go to the other extreme and ignore the extremely valuable information which can be gathered from such measurements. At the BBC, loudspeakers are, in the end, judged subjectively on their ability to reproduce program material accurately, not just as a pleasant sound, and are judged by comparing the reproduced quality with that in the studio itself. When, however, questions of the basic design or modifications are involved, it is found that these can usually be determined simply by objective measurements in a free-field room. This paper describes the hows, wheres, and whys of the tests made during the development of BBC monitoring loudspeakers. The order in which the items are given is not to be taken as an indication of their importance.

Frequency Response

The steady state axial frequency response characteristic test is carried out by measuring the axial sound output as a continuous function of frequency, at a specified distance from the loudspeaker, in free-field surroundings when a constant a.c. voltage is applied to the loudspeaker terminals. It is the measurement which is most often made and contains a great deal of information.

There have been suggestions that since a listening room clearly departs widely from free-field conditions that the loudspeaker output should be measured in a live room. It is assumed that because a listener usually sits sufficiently far from the loudspeaker to be largely in the reverberant sound field that this is the factor which should be measured. In fact, the ear does not take account of the reverberation as a first order quantity but only as a second order, otherwise a person speaking in one room would sound quite different when in another room having differing characteristics, and we know from experience that this is not the case. In practical conditions the ear fastens on the direct sound and although the reverberation cannot be neglected, relegates it to a secondary place. Measurements taken under specified free-field conditions therefore contain much more relevant and easily interpreted information than those taken under live conditions which apply to that room only.

Another suggestion [1] that has been made is that an intermediate condition should be used and measurements should be made with the loudspeaker radiating into an infinite plane, i.e. into 180 degrees instead of 360 degrees. Compared with free-field measurements this would give a bass lift to the response up to a frequency which would depend on the size of the cabinet. This bass lift would therefore be a variable quantity not easily allowed for; furthermore it is admitted in the same article [1] that we do not in practice hear such a bass lift and the free-field measurement seems to agree best with what is heard in a practical situation.

The test conditions for the steady state axial frequency response characteristic need therefore to be specified quite closely.

In the first place true free-field conditions are assumed for most cases, that is unless a loudspeaker is designed to be mounted in a corner or so that the sound is deliberately reflected from a wall or ceiling. True free-field conditions can only be obtained in the open air at least 30 feet from any obstacle or in a large enough free-field room. In the latter case, the author has shown elsewhere [2] that it is necessary for the tips of the wedges on opposite sides of a free-field room to be at least one wavelength apart at the lowest frequency of interest for free-field conditions to apply, even with perfect absorption at the wedges. The trouble is that excess absorption takes place, as in an acoustically lined duct, when the spacing is appreciably closer than this; with too small a room this will have the effect of giving an apparent bass cut. In the larger free-field room at the BBC's Research Department a special type of polyurethane wedge is used and the dimensions are such that free-field conditions exist to below 40 Hz within ± 1 dB out to 10 feet from the loudspeaker under test [3].

In addition to providing free-field conditions it is essential for the measurement of the axial frequency response to be made at an adequate distance from the loudspeaker, particularly for multiple unit designs. A minimum distance of five feet is adopted for this sort of work, for it can easily be calculated that at closer distances the relative contributions of l.f., m.f. or h.f. units is changed significantly and a wrong appreciation will be obtained of what the listener will hear, in practice, at a distance of 6½ feet or over.

The next question is that of the bandwidth to strive for. We can adopt the rather näive approach that as the ear can hear frequencies over a range of 16 Hz to 20 kHz, or to over 30 kHz for children, we should aim for this range with all its attendant difficulties. At the BBC, however, we have adopted the rather more mature engineering approach of trying to determine the narrowest bandwidth which can be used

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without the listener noticing any degradation in quality. In a series of experiments [4], known under the delightful name of Operation Clothear, the upper cut-off frequency of program material was altered and the number of persons who could detect the change on an ABA test was found. The program material was carefully selected to be the most sensitive for this sort of test and observers whose ears had been checked were used. Even under these very critical conditions, surprisingly few observers were able to detect a cut-off frequency of 12 kHz. As a result it has been decided that monitoring loudspeakers should have a response extending to at least this frequency and that if this can be achieved on the axis, greater weight should be attached to obtaining, (a) a good spatial distribution, (b) a smooth curve and, above all, (c) a high degree of repeatability, than to extending the frequency of cut-off.

At the bass end the decision is more difficult and as an engineering compromise between size, cost, and response, the latter is maintained to about 45 Hz and allowed to fall below this figure.

It should be made clear that whilst the axial frequency response characteristic is a necessary measurement, it is by no means sufficient to obtain a smooth or even flat response curve. Very little work has been done to determine either the smallest irregularity which is audible, how wide-range trends in response affect the reproduction, or even, given a perfectly smooth axial frequency response curve, whether it should be flat to give the most faithful reproduction. Although it is often assumed to be true, it is doubtful whether a flat axial response curve gives the most realistic performance, but in this connection it is necessary to state our own assumptions. At the BBC we assume that the microphone and the amplifiers should have a uniform response; for tests on new types of loudspeakers the microphones used are equalized to be uniform ± ½ dB over the frequency range of 40 Hz to 15 kHz, or beyond if it is possible to do so without degrading the signal-to-noise ratio too much. It then follows that for the most realistic performance the axial frequency response characteristic of the loudspeaker must be allowed to take any form dictated by the ear, and it is found in practice that a slight slope over the frequency range from 200 Hz to 5 kHz is desirable, the response at the latter frequency being about 3 dB lower than the former. It should not be surprising that a uniform curve is not ideal, for the sound field in the listening room is very different from that in the studio and if, psychologically, a trend in the axial frequency response characteristic gives a better illusion of realism, this is regarded as entirely justified. There is also the factor that the aural effect of small degrees of coloration can be reduced by "cooking the curve." This procedure must be used with care, however, as it is not rigorous and it can easily be overdone.

It should be noted that the ear is not uniformly sensitive to broad-band changes throughout the frequency range. Thus a change in level in the 500 Hz to 2 kHz band of 1 dB is audible and one of 2 dB is quite marked. On the other hand, at the extremes of the range a change of 2 dB is barely audible at all.

Some figures from our experience are worth recording here. From the point of view of local irregularities we have an octave-band variable equalizer which in the "flat" condition shows a ripple on the frequency response curve of ± 1½ dB. That equalizer can be switched in or out and it can be stated definitely that this degree of ripple in a flat average response is absolutely inaudible. On the other hand we have had a case where a microphone had a smooth downward slope of 3 dB over the range of 100 Hz to 3 kHz. This was detected and equalized by ear by the program operators to within ± 1½ dB without reference to any kind of objective measurement! The obvious moral is that small local irregularities are permissible and that there is little point in aiming at too smooth a curve, but that broad trends are detectable to quite a fine degree.

Off-Axis Response Characteristics

The off-axis response is measured in a similar manner to the axial characteristic and is important for two reasons. Firstly, we do not always listen to a loudspeaker whilst seated on its axis, and secondly, it is largely the off-axis curves which determine the reverberent sound.

Taking the first point, it is important with monitoring loudspeakers, and to a lesser extent with the domestic types, that there should be a wide angle over which a listener can hear accurate reproduction, preferably indistinguishable from that on the axis. With multi-unit loudspeakers, apart from the coaxial types, this implies that care must be taken in mounting the units to get the best distribution in the desired plane. Thus for normal monitoring and domestic listening a two-unit loudspeaker would have the units mounted one above the other so that the system is symmetrical in the more important horizontal plane. In some cases in broadcasting, e.g. in a mobile control room, the opposite may be the case and it may be in the vertical plane that uniform characteristics are required. A further limitation with multi-unit loudspeakers is that there is a miminum distance at which they should be listened to if equal contributions from the units are expected.

The sort of trouble that is experienced off axis with a two-unit loudspeaker is illustrated in Fig. 1. The two units might, for example, be a 12 in. woofer and a two in. tweeter. If the overall response is made flat on the axis, that at 60 degrees might well follow the second curve, for at the upper end of its band the woofer could be quite directional whilst the tweeter, where it takes over, should be omnidirectional. This variation can be reduced by partially covering the woofer with plates leaving only a narrow slit to radiate the sound. The process must not be carried too far however, as the inertance of the slit resonates with the compliance of the air inside the cone giving a peak in the response followed by a sharply falling response. The degree of improvement effected by the slit never reaches the full theoretical amount; this is discussed in greater length in Ref. 5.

Greater uniformity in response with angle can of course be achieved, at a cost, by employing three units each covering a narrower frequency range. By judicious use of these methods the off-axis curves can be smooth and follow that on the axis within ±3 dB for angles up to 60 degrees over most of the frequency range.

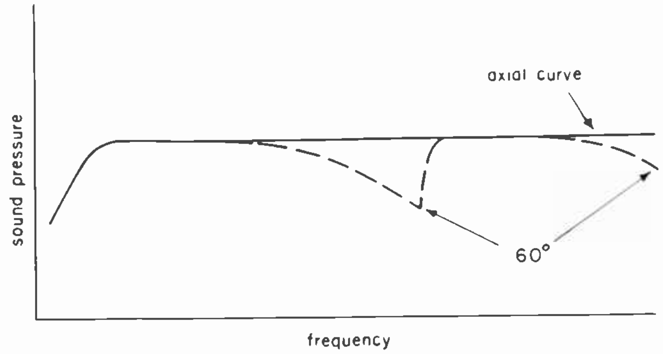

Fig. 1—Two-unit loudspeaker. Nominal frequency response characteristics on axis and at 60 degrees in the plane at right angles to that containing the two units.

The response in a plane containing the units is also irregular as at some angle in the crossover frequency range the two units are half a wavelength apart and a cancellation occurs at smaller angles than in the orthogonal plane, as shown in Fig. 2. Fortunately the crevice is narrow and the frequency varies at a discrete frequency; it does however mean that appreciable off-axis angles in this plane are to be avoided if possible. This means that contrary to many advertisements, multi-unit loudspeakers have definitely a "right way up" for serious listening.

The influence of the off-axis response on the reverberation is of course very large. If the loudspeaker is regarded as the center of a sphere which is divided into concentric bands occupying equal angles at the center, then the area covered by $\pm$ 5 degrees say will only occupy a small fraction of the area covered by 85 degrees $\pm$ 5 degrees, and the contribution to the total energy radiated into the room will be correspondingly small. The reverberation will thus be largely determined by the off-axis curves and it is at once apparent that any large discrepancy between direct and reverberant sound will be detected.

Polar Response

For this measurement the loudspeaker is mounted in the free-field room, and the measuring microphone rotated about it by a boom controlled by selsyn motors from outside the room and which also control the rotation of the polar recording paper. As with the axial frequency response characteristic, it is essential to provide true free-field conditions and the microphone must be at a distance from the loudspeaker great enough to give representative results, say five to six feet. Measurements are taken either at discrete frequencies or, more usually, employ bands of noise when general trends are required.

The polar response is of course another way of regarding the off-axis curves discussed above. It is not used extensively however because it is not the polar response as such which is listened to but the frequency response characteristic at a specific angle. The polar response measurements are therefore largely used to supplement the response at angles when a specific feature is to be examined at one particular frequency or band of frequencies during the design of the loudspeakers.

It is also useful in estimating the service area which will be well covered by one loudspeaker or in calculating the directivity or total power radiated by the loudspeaker.

Directivity and Power Response

The directivity of a loudspeaker is a measure of the degree to which a loudspeaker fails to be omnidirectional and is defined as the total acoustic power radiated at a frequency, or band of frequencies, compared with the power which would be radiated by an omnidirectional source having the same axial output. When measured in bands over the whole frequency range, it gives an indication of the way the reverberant sound will differ from the direct sound heard on the axis for a nominally flat axial frequency response characteristic and the two are therefore best dealt with together.

Since the parameter we want determines the reverberation level, this at once gives a clue as to one method of measurement. The loudspeaker is stimulated with bands of noise and the reverberant field measured as a function of frequency. By knowing the absorption characteristics of the room, the total radiated power can then be calculated from the formula:

$$\text{SPL} = \text{PWL} + 10 \log_{10}\left(\frac{Q}{4\pi r^2} + \frac{4}{R}\right) + 0.5\ \text{dB}$$

Where SPL is the sound pressure level re $2 \times 10^{-5}\ \text{N} \div \text{m}^2$ PWL is the power level, Q is the directivity factor, r is the distance in feet from the loudspeaker to the microphone and R is $S\alpha \div 1 - \alpha$ where α is the average sound absorption coefficient for the room and S is the area of the bounding surfaces of the room in square feet. In practice a reverberation room is used as this gives a more uniform field and has known absorption characteristics. However, similar limitations as to size apply to this room as to the free-field room and unless the room is large enough, true integration will not take place at the bass and in addition there is always some danger of the vent resonance in a vented cabinet being affected. It is however the most widely used method and properly instrumented, taking measurements at a number of points in the reverberant field, can give fairly accurate results.

The directivity can also be obtained in a free-field room by recording the polar radiation pattern at a large number of angles around the loudspeaker and calculating thence the directivity. As these measurements must be carried out at a number of frequencies, the labor involved is quite large and this method is rarely used.

The method employed at the BBC is similar but more convenient and quicker, the details being described in Ref. 6. In practice it consists of integrating the total power output "ρ" of a microphone as it is rotated around the loudspeaker in

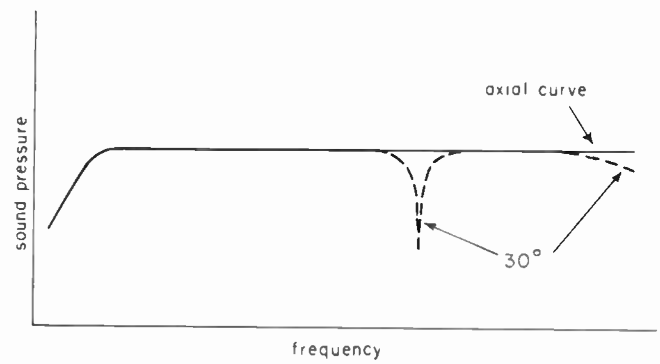

Fig. 2—Two-unit loudspeaker. Nominal frequency response on axis and at 30 degrees in the plane containing the two units.

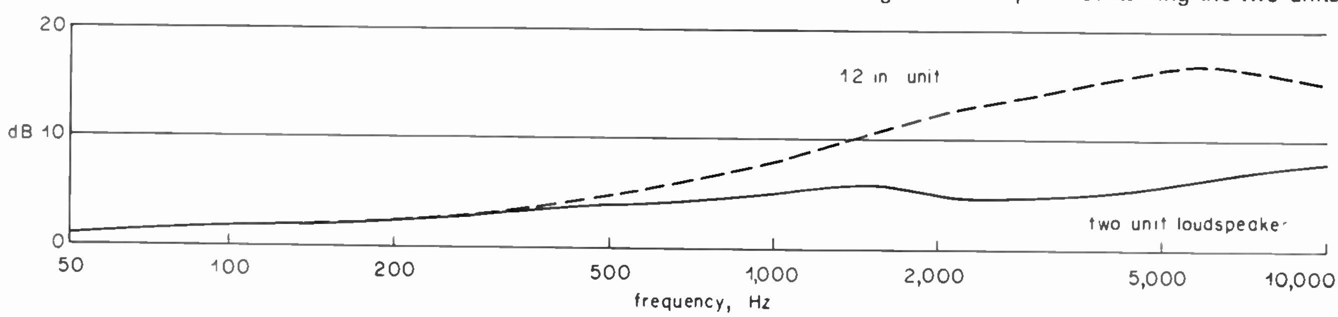

Fig. 3—Directivity of a single 12 in. unit and of a two-unit monitoring loudspeaker having a 7 in. slot in front of the 12 in. bass unit.

the free-field room in sectors, rather like the segments of an orange, for which the integral to be determined is

$$\int_{0}^{\pi} \pi\rho^{2} \quad \mathrm{Sin}\,\theta \times d\theta$$

The microphone is fed to a sine law potentiometer and to an integrator so that the directivity can be measured for any frequency. Since the free-field room is usable down to 40 Hz, the directivity can be measured over the whole spectrum without difficulty.

An illustration of the sort of result obtained is given in Fig. 3, both for a simple 12 in. radiator and for a two-unit monitoring loudspeaker.

It will be noted that the curve of the directivity of the latter, although much more uniform than that for a single 12 in. unit is still not flat. In the nature of things a 3 dB slope is to be expected as the bass unit is fundamentally omnidirectional whilst the tweeter can at best only radiate into a hemisphere. It is clear therefore that even if found to be desirable, a loudspeaker having a flat total power response cannot be achieved using a conventional cabinet. It is of interest to note here that a monitoring loudspeaker with a close approach to an omnidirectional middle and high frequency unit was designed by one broadcasting authority [7] but later designs from the same authority have retreated considerably from this concept. Our tests on this loudspeaker with speech and solo instruments certainly indicated that the directivity was too small for this type of program material and the later changes by the designers indicate an acceptance of this verdict. For example, with speech, too great a degree of diffusion will give the impression of a voice spread over a large area. On the other hand, at the BBC with more conventional types of monitoring loudspeaker, any increase in angle of radiation so far has been welcomed. There is therefore some sort of optimum which, however, has never been satisfactorily determined, and measurements such as the total power response for differing types of loudspeakers will help to settle this feature in the future.

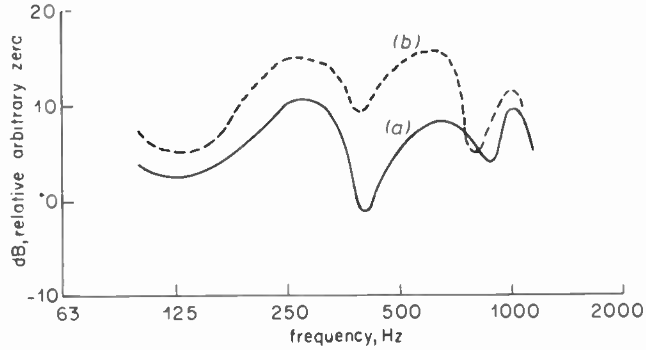

Fig. 4—Curve a, measured frequency response characteristic of loudspeaker in a corner using warble tone. Curve b, frequency response characteristic of loudspeaker in a corner calculated from three images in walls and ceilings. (Curves arbitrarily displaced.)

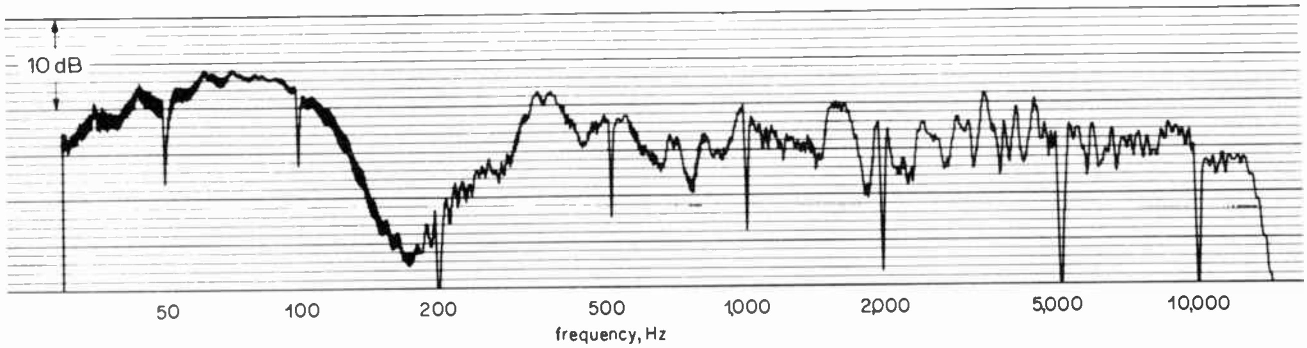

Fig. 5—Measured frequency response characteristic of a loudspeaker when touching the three surfaces in a corner.

Corner Mounting

It is sometimes most convenient to mount a loudspeaker of the conventional cabinet type in a corner. This may be to try to narrow the angle of the area to be covered or simply to hang the loudspeaker out of the way of the general impedimenta in the room. At the BBC this has been carried out particularly in television control rooms where the monitoring loudspeaker has been hung over the television monitors which are placed in a corner.

However, as the quality of speakers has improved there has been increasing dissatisfaction with the quality of reproduction from a corner placement and complaints of coloration have been made which do not apply when the same loudspeaker stands free of the corner. Measurements of the output of a speaker, fed with warble tone to remove standing waves in the room, have been carried out *in situ* with the results shown in Fig. 4, curve a. On the assumption that the irregular response was due to interference between the various images formed in the adjacent walls and ceiling, curve b shows the expected response. It will be seen that the two curves are very similar and it is not surprising that coloration was noticed at a frequency just below 300 Hz. The effects of a corner position can be mitigated by asymmetrical mounting and also by the use of absorbing materials in the corner, but these are palliatives and the use of corners for normal speakers is to be avoided whenever possible. The effect on the frequency response characteristic of placing the loudspeaker right in the corner is shown in Fig. 5 as an awful warning! For further details of these tests see Ref. 8.

Distortion and Overload

This is a subject on which most authors are silent, though not without reason. Total harmonic distortion figures of small fractions of one percent are gaily quoted by amplifier and equipment manufacturers and are expected by the customers, but figures as to the distortion generated by the loudspeaker and therefore actually heard, are few and far between. The problem divides itself into two parts, the difficulty of making meaningful measurements and the interpretation of the results.

A loudspeaker has a number of sources of distortion, viz. voice coil amplitude, spider, surround, and of course, the cone. The latter can be regarded as a transmission line, open circuited at one end and only roughly terminated at the other, having differing velocities of propagation in the radial and circumferential directions. In the latter case the fundamental frequency for a straight sided 12 in. cone will be between 50 and 100 Hz with frequent overtones above this. Radial modes do not usually set in before 400 Hz but the surround can cause trouble in this frequency region too. Even the spider will resonate and have standing waves causing an irregular frequency distortion curve, and the only item which has a smooth curve in this respect is the voice coil-magnetic field system.

It is therefore not surprising that the frequency distortion curves are extremely irregular, much more so than those of the fundamental. In order to obtain meaningful results, therefore, it is even more essential than it is for the fundamental to employ a method of measurement giving the various orders of distortion as a continuous function of frequency.

There are three such methods of measurement available. One due to Olsen and Pennie [9] employs a series of high pass filters which are switched in automatically as the test frequency is increased so removing the fundamental and allowing the sum of all harmonics and noise to be measured. Although better than nothing, it will be shown later that this measurement of total harmonic distortion is not very meaningful.

The second method is due to Bruel and Kjaer who use their 1/3rd octave band-pass filters, again switched in automatically, to measure the second and third harmonics as a function of frequency. This is better but of course we would very much like information on the higher harmonics, which is not possible with this set-up owing to the comparatively wide bandwidth of the filters. What we really need is an instrument which measures harmonic distortion up to about the eighth order as a continuous function of frequency. Since, by definition, this order harmonic cannot be measured at a higher fundamental frequency than three octaves below the upper cut-off frequency of the loudspeaker, these curves should be supplemented by intermodulation tests, again as a continuous function of frequency, which of course can extend right up to the cut-off frequency itself. Since no such instrument was available one was designed by the author for use at the BBC [10]. This is not the place to enter into details of its design, which is described in the reference given, but by means of heterodyne methods, this versatile instrument enables both harmonic and intermodulation distortion curves to be taken as described above. It is a pity that although the patent is available for exploitation, no instrument firm has produced it for use by other organizations. A typical set of curves is given in Figs. 6 and 7 for a high quality monitoring loudspeaker taken at a sound level of $1N \div m^2$ at five feet in a free-field room. Note not only how low the average distortion is but also that the higher order distortion curves are very irregular and that the frequency at which one harmonic is a maximum may even by a minimum for another. For example, if we look at the difference between the sixth and the eighth harmonics at 55 and 59 Hz, at the lower frequency the eighth is at least 22 dB above the sixth, whilst at the higher frequency it is 19 dB below, a relative change of at least 40 dB in 4 Hz! Between 250 and 260 Hz, there is a corresponding difference of over 25 dB. In fact the figures are even greater than these but the curves have been cut off at -90 dB as they cannot be guaranteed below this level.

The intermodulation curves are comparatively smooth in this case as they largely relate to the tweeter which in this design moves almost as a rigid piston up to the highest frequencies, and therefore does not break up into resonance modes.

The interpretation of these curves needs some care. In the first place, although we can see that the general trend of the curves for such a high quality loudspeaker is smooth, on the other hand, because of the irregular detail as described above, it is not possible to get the average separation of the curves by means of measurement at a few spot frequencies. The next most important point is that a simple rms sum of the levels is quite inadequate. No one would seriously dispute that one percent of seventh harmonic is far more objectionable than the same level of second harmonic. As long ago as 1937 it was demonstrated by the R.M.A. [11] that to get a reasonable subjective assessment, the level of the harmonics should be weighted at least according to their order. Since then two papers [12, 13] have clearly indicated that the weighting should be according to the square of the order, that is, instead of using the rms sum of the harmonics, each harmonic level is multiplied by $n^2 \div 4$, where n is the order of the harmonic, before taking the rms sum. In this way the level of the second harmonic remains unchanged. It is the need for this type of weighting which shows the inadequacy of the simple rms figure measured by the first of the tests described earlier.

It should be noted here that some nonlinearities can be highly nonlinear, that is to say that they may even increase rapidly with input level and then decrease again as a percentage of the fundamental. The surround is particularly susceptible to this, both near the half wave resonance point and in the bass. In the first case owing to resonance, the amplitude may increase rapidly with increasing input until the highly nonlinear region is reached. Distortion is then at a maximum and cannot increase. However, as the input voltage is increased, the output from the cone will still increase and the total percentage distortion will therefore be reduced. A similar case occurs at the bass end, particularly at the vent resonance frequency of a vented cabinet. Here, when the cone moves, say, inwards there is a very high back pressure in the cabinet pushing the surround outwards and, if it is very compliant, the surround may actually move in the opposite direction to the cone until the elastic limit is reached. Thus it will execute almost a square wave in antiphase with the cone, but again as the input power is increased the total distortion will reach a

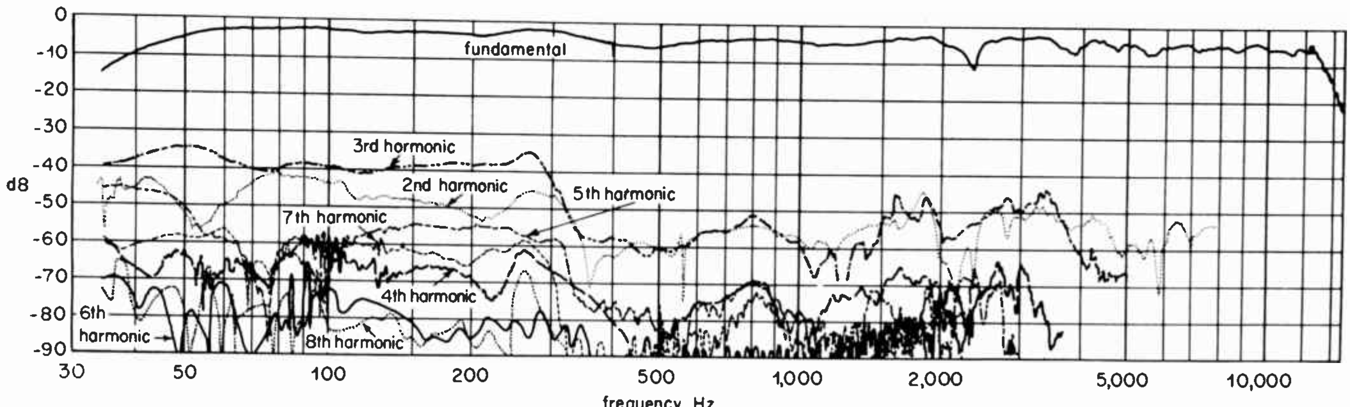

Fig. 6—Harmonic distortion curves of three-unit monitoring loudspeaker in free-field room. Sound level $1N \div m^2$ at five feet from loudspeaker.

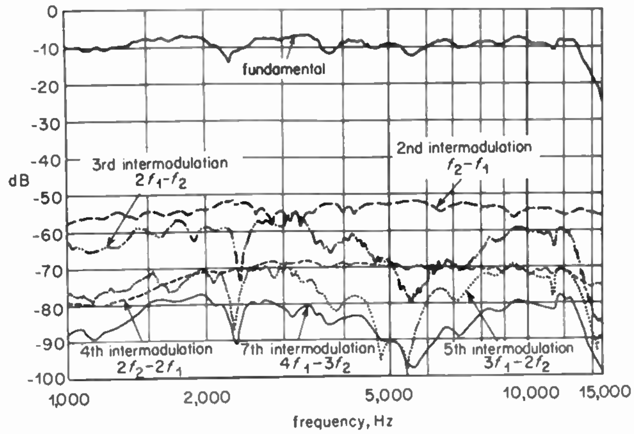

Fig. 7—Intermodulation distortion curves of three-unit monitoring loudspeaker in free-field room. Sound level $1\,N \div m^2$ at five feet from loudspeaker.

peak and then be reduced. Note that in each case high orders of harmonics may well be produced in the process and that the maximum distortion may actually be at low or medium sound levels.

The overload level is related to the distortion level in a complex way. The two variables are the peak to rms ratio of the program and the spectrum concerned, as in practice amplifier/loudspeaker systems overload on peaks well before they burn out. For example solo piano will overload loudspeaker systems at much lower loudness than will an orchestra, and organ pedal notes will show up any excessive bass equalization. Thus to arrive at a stable figure, we use bands of pink noise. It may seem surprising that the overload point of noise can be heard in view of the nature of the spectrum but in practice it can be determined by ear within $\pm$ 1 dB.

Transient Response

The transient response of a loudspeaker can be measured by placing it in a free-field room and determining the response to a sudden impulse such as a square wave or by the response to bursts of tone. In theory the former test contains all the desired information but in practice it is difficult to analyze, particularly because, as will be shown later, it is necessary to measure transients well below the steady state level.

In practice therefore only the chopped tone method is useful. In this test the input to the loudspeaker is gradually changed in frequency whilst the amplitude is chopped at the input of the power amplifier (so maintaining the correct damping at the terminals of the loudspeaker) at a rate of about five times per second, so that the burst of tone lasts for about 100 mS and the off period for similar length of time. The repetition rate is a cross between a high value allowing a rapid frequency glide and a slow enough rate to allow steady state conditions to be established. For very high Qs even slower repetition rates are necessary. During the off period the output of the loudspeaker is examined for resonance which will show up as a "tail" on an oscilloscope. The degree to which the level of the commencement of the tail is below the steady state is measured; this is known as the dilution of the resonance. The Q and the frequency of resonance are also noted. At one time it was customary, at the BBC, to take delayed response curves, that is curves of the output from the loudspeaker at intervals of 5, 10, 20, 30, etc. mS after the tone had been cut off. This gives a very good picture of the transient response but is a rather lengthy procedure and the present practice is merely to glide throughout the frequency range noting the parameters given above.

The measurement of transients is another aspect of loudspeaker testing which reveals our ignorance on the subjective side. The importance of the transient response generally seems to be badly underestimated for it is no exaggeration to say that with modern high quality units the coloration caused by a poor transient response is the main factor in determining the sound quality of the loudspeaker. A good example is shown in Fig. 8. This shows the axial response of two loudspeakers of similar size, and as a matter of interest, designed by the same engineer when he was at two different firms. The top curve shows the axial frequency response curve of his first design and the lower curve his second, both curves taken by the present author. The progress made in smoothing out the axial response is commendable but the awful fact is that the first loudspeaker sounds very much better than the second. This latter has severe coloration centered around 500 Hz just where it will be seen that the axial response curve is specially smooth, whereas the irregularity in the upper curve near this frequency is relatively innocuous. This amount can be capped by the behavior of a middle frequency unit designed by us for a three-way monitoring system [14] and which also had a nicely smooth axial frequency response characteristic. On completion of the loudspeaker, listening tests showed a marked coloration in the 1500 Hz region even though the middle frequency unit had passed our usual tests. Still more careful measurement with chopped tone, however, showed up three resonances close together in frequency which had a dilution of no less than 40 dB, but a Q of about 500! Two things are noteworthy here. Firstly the effect of such resonances on the steady state is only 0.1 dB peak if they are in phase with the steady state

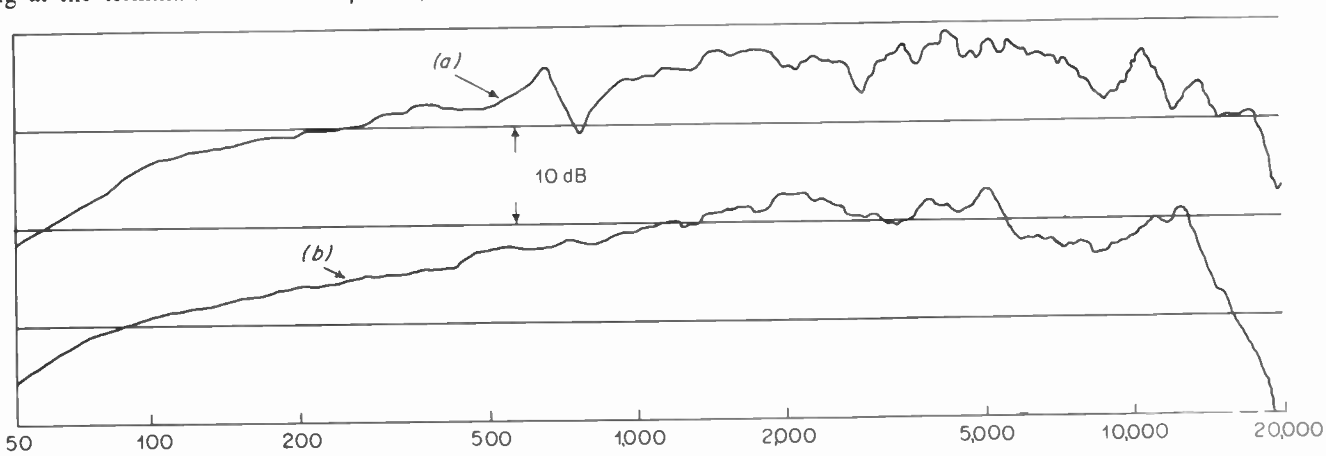

Fig. 8—Axial frequency response curves of two small two-unit loudspeakers; (a) first design, (b) second design. (Curves arbitrarily displaced.)

response and very much less still if th. y are in quadrature. Secondly it is rather surprising that a material which in flat sheet form has a Q of much less than 1 should, when formed into a hyperbolic cone, have such a high Q. It is clear that a smooth steady state curve, whilst obviously desirable, is not in itself a guarantee of absence of coloration. It is here that the German DIN standard No. 45 500 falls down. In an attempt to define conditions necessary for a good loudspeaker from the transient aspect, a maximum slope is laid down of 12 dB per octave over any portion of the axial frequency response curve. Whilst, incidentally, this rules out a vented cabinet with its bass slope of 18 dB per octave, it is quite impossible to cope with the case cited above, of a maximum disturbance of 0.1 dB, in these terms. Whilst theoretically the information is indeed in the steady state curve, in fact it is too oblique a measurement of this parameter to be useful and the chopped tone technique is the only possible approach.

Several attempts have been made to record automatically the energy in the tail of the transient but in practice the required parameter is by no means clear. Experiments to determine the subjective correlation between frequency, Q, phase, and dilution are at present being conducted here but an indication of the difficulties in the way of instrumentation lies in the fact that subjectively a suppressed zero seems to be involved and preliminary results suggest that two resonance close together in frequency may add, not on an rms basis, i.e. 3 dB as do pure tones, or even arithmetically, i.e. 6 dB, but possibly to the tune of 10 dB. At the moment therefore we cannot predict the effect of a resonance; all we can do is to listen to program material or to pink noise through the loudspeaker, measure with chopped tone those resonances in the vicinity of a coloration, and increase either the dilution or damping or both until the resonance is inaudible. In the meantime it is clear we must examine at least 50 dB below the steady state and look for Qs up to 500 or more. This calls for refined conditions of measurement, particularly in terms of standing waves in the free-field room.

Phase Frequency Response Characteristics

With few exceptions [15] the phase frequency response characteristics of loudspeakers are usually regarded as unimportant, and this accords with our experience. Measurements of phase response have been made here with the test loudspeaker in the free-field room by employing a wide range capacitor test microphone, a delay line and a phase meter of the zero crossing type connected to a level recorder. Except in the region of crossover such measurements have not been found to provide useful data and even in this restricted case equally useful information can be obtained by observing the individual contribution of the two units concerned and the way they add together.

One organization does go as far as to displace the various loudspeaker units one behind the other in order to be able to reproduce a square wave well, but it should be noted that this will only apply on the axis and leads to a complicated expensive cabinet system. We have found that by approximately adjusting the crossover network, the outputs can be made to add simply even when the units are in the same plane so that it is impossible to detect from the axial frequency response curve at what frequency the crossover is. Furthermore this will hold over the whole horizontal plane containing the axis. We have not found any further attention to phase to be necessary.

Doppler Effect

This falls into a similar category to the effects of phase in that while it must exist, we have never been able in practice to attribute any ill effects to this cause. This may partly be due to the fact that all serious listening at the BBC is done on loudspeakers with at least two units, and this of course will greatly reduce the Doppler effect. Even however with such wide range single-unit loudspeakers as the author has examined, it can be said that other faults have at least been far more important, but it is of course possible that with further progress the Doppler effect will become noticeable on program material as a small residual. No tests are therefore made for this effect.

Subjective Testing

This is the touchstone and none of our previous work is adequate if this test fails. It may be asked how this is possible in view of all the measurements we have taken, and some indications have been given in the sections concerned but it will not hurt to repeat them here. To start with, for a monitoring loudspeaker the quality of reproduction must be that of the original in all its stark reality, with no pandering to a "pleasant sound." In this it is assumed that we start with a microphone having a perfectly flat frequency response curve. But in spite of this we are still not sure what the optimum frequency response characteristic of the loudspeaker is, how much coloration we can stand, or what the best directivity is. Since the final result is subjective, we can only determine these conditions by subjective experiments and then lay down the objective results. Finally for a monitoring loudspeaker the results must hold for any type of program material. Thus a loudspeaker which obtains a very high degree of diffusion pleasant for reproducing an orchestra will not do for speech if a commentator appears to have a mouth six feet wide! The desired listening conditions must also be laid down. For a broadcasting organization it is assumed that the majority of listeners will be in their own homes, probably, for serious listening, in a living room. To this end a very large number of measurements have been made in listeners' homes and an average reverberation time of 0.4 seconds arrived at [15]. Listening rooms are therefore made to have this value of reverberation whenever possible.

One of the best forms of test material is also, strangely enough, the easiest to obtain, that of well known male voices speaking from dead surroundings. It is a fact that we are particularly sensitive to nuances in the human voice, a vast number of differing voices can be distinguished, and a well known voice is excellent test material. It has often been observed here that a loudspeaker which is balanced to reproduce the male voice is also excellent, over this frequency range, on music and other types of program material while the reverse does not necessarily hold at all. A further advantage of the speech test is that the person whose voice is being reproduced can stand behind the loudspeaker concerned and alternate live with reproduced speech.

For music tests it is necessary to have a studio at one's disposal together with an adjacent listening room, and to listen to a wide range of instruments, solo as well as in a full orchestra. Furthermore it is essential to use a single microphone pickup rather than multimike technique, or else it is not possible to listen directly to what the microphone is picking up. Recordings are a poor second best to a real performance as it is not possible to know the microphone characteristic, the reverberation, or even how the orchestra was playing that day. The latter point is quite important, as on one occasion, for example, the author thought the sound of the violins rather harsh over an experimental loudspeaker and was very relieved on entering the studio to find that harshly was exactly how they were playing at that moment.

Finally for outside broadcasts the listening conditions are often far from ideal and the loudspeaker has to be able to cope with these too. Generally the fault with such conditions is that there is not enough acoustic treatment present, a trend which is also becoming apparent in some modern homes, where the old type of deeply cushioned furniture is replaced with more sparcely upholstered types. In such circumstances the sound tends to be harsh and so will emphasize any such tendency in the loudspeaker. It is a truism to say that any excess is objected to more than a corresponding degree of deficiency. The experimental loudspeaker is therefore sent on a field trial under differing listening conditions and with differing studios.

It should be mentioned under this heading that one very convenient form of subjective test when an alteration is to be carried out is to make an instantaneous changeover between the two conditions. Now this is not always possible directly, as for example when the amount of damping compound on the cone is to be changed. It is not usually satisfactory to have one loudspeaker in one condition and one in another, as generally the difference between loudspeakers even of the same type is audible. The method we have employed is to record test material such as pink noise on the axis at a specified distance in the free-field room under each condition, one on either track of a two-track tape recorder. On replay on a monitor, switching between the two tracks can be instantaneous and the effect of even small changes can be made quite obvious and a record of them held.

Tolerances

When the design of a loudspeaker is fixed the only three parameters likely to change in production are the frequency response characteristic, the crossover network, and the coloration. Other factors such as the directivity, overload, etc., are usually constant. For a monitoring loudspeaker one essential goal is that all units should sound alike, so that if a producer records program material in one studio using one loudspeaker and edits the tape elsewhere, the balance should be identical. As all makers of loudspeakers know only too well, such a condition is extremely hard to achieve and until recently would have been thought impossible. The tolerances which the user will fix will therefore be tight as possible but at the same time must be realistic or no loudspeakers will pass the test. The response of paper pulp cones has been notoriously difficult to control in the past, particularly the thinner cones, for after all they are merely an exercise in statistics with the pulp fibers as the variable! The position has been radically improved with the use of vacuum-formed thermoplastic materials. With the right materials these can be exceptionally free from coloration and give repeatable frequency response characteristics.

This leaves the crossover network components as the remaining variable. In order to obtain the sort of frequency response required of monitoring loudspeakers, crossover networks for the last quarter of a century in BBC designs also act as equalizers for the units themselves. The tolerance on components for these two purposes is fixed at $\pm$ 2 percent to maintain monitoring standards, and for this reason paper dielectric capacitors and gapped mu metal-cored inductors must be used to maintain the required stability over a long period.

For the studio monitoring loudspeaker type LS 5/5 the tolerance over the whole frequency band for the general trend of the frequency response characteristic is $\pm$ 1 dB with respect to the standard laid down. A further small allowance is made for minor local deviations from this standard. It is found to be much more satisfactory to divide the tolerance in this manner than if, for example, the two tolerances were added to give, say $\pm$ 2 dB overall instead. This would allow larger deviations in the general trend, which is not desirable.

As may be expected from these tolerances, the degree of uniformity of performance is very good. It is rarely possible to be able to detect differences between loudspeakers even on a direct changeover and certainly not by walking between differing studios. It also means that any two loudspeakers can be used as a stereo pair and provide an excellent sharp image.

It is of some satisfaction that we can state that the first production batch of these loudspeakers all passed this stringent test without any failures thus indicating the degree of precision now possible in the loudspeaker field. For comparison it should be noted that the tolerances on the frequency response characteristic of the best grade of capacitor microphones is 5 dB at the middle and high frequencies and 7 dB at the bass. With careful design, monitoring loudspeakers can at last be regarded as precision instruments.

Conclusions

It has been shown that many objective measurements are necessary during the design and testing of loudspeakers and it is true to say that the time has passed when a high quality loudspeaker could be constructed without their aid. On the other hand there is still a good deal of ignorance as to the exact design goal defined in objective terms, and further research should be carried out to elucidate these items, especially in the field of coloration. In the absence of such information we still have to fall back on a subjective test as the final assessment. Æ

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Activating Your Loudspeaker Crossover

Michael W. King

ONE OF THE PROBLEMS in hi-fidelity system design is that of a crossover network to divide the output spectrum into portions suitable for high and low frequency loudspeaker drivers. In the past, this has been achieved with the use of passive filters, as shown in Figure 1A. The disadvantage of this system is the non-constant input impedance of a direct-radiator loudspeaker.[1] A solution to this problem has been proposed by Ashley and Kaminsky,[2] employing a system like that shown in Figure 1B. In this system filtering is accomplished at low levels and the speakers are driven by low output impedance buffer amplifiers. Clearly, this isolates the driving point impedance of the loudspeakers from the output of the filter; and, in addition, the input impedance of the buffer amplifier can be made virtually any value of pure and constant resistance making this system also suitable for use with passive filters. In addition, it should also be noted that the typical passive network of Figure 1A requires capacitances on the order of tens of microfarads, which are expensive and difficult to obtain, while the active network of Figure 1B requires capacitors on the order of nanofarads, reducing the cost differential between the two systems.

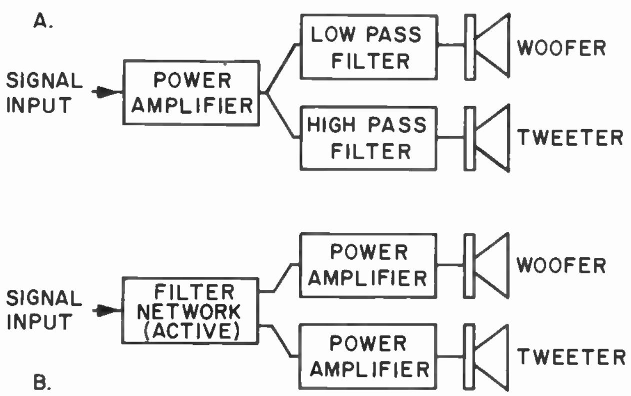

Fig. 1—Possible crossover-amplifier arrangements.

Following Ashley and Kaminsky,[2] a filter as shown in Figure 2 was constructed. This is an asymmetrical third-order network where the high-pass has a third-order Butterworth characteristic. Here the high-pass is achieved by an active filter and the low-pass portion is derived by subtraction of the high-pass signal from the total input signal. This design is of the constant voltage type described by Ashley[3] and Small,[4] meaning that the phasor sum of the outputs of the filter is equal to some constant in the frequency domain. Because of this, given perfect drivers and amplifiers, the total acoustic output signal will be a perfect reproduction of the source material. This can be demonstrated, as shown below by adding the transfer functions of the high and low pass filters to give the transfer function for the sum of the outputs.

$$\frac{e_{total}}{e_{IN}} = \frac{e_{HI}}{e_{IN}} + \frac{e_{LO}}{e_{IN}} = \frac{-0.5\,S^3}{S^3 + 2S^2 + 2S + 1} + \frac{-0.5\,(2S^2 + 2S + 1)}{S^3 + 2S^2 + 2S + 1}$$

$$= -0.5\,\frac{S^3 + 2S^2 + 2S + 1}{S^3 + 2S^2 + 2S + 1}$$

$$= -\frac{1}{2}$$

Thus, this network has a gain of one-half which is constant with respect to frequency. The filter was designed for a gain of one-half in order to simplify component values and to avoid problems caused by excessive levels.

A μA741 operational amplifier was chosen because of its high gain and wide range of operating voltages. This was chosen over the popular μA709 operational amplifier because the 741 requires no external frequency compensation. Supply voltage of $\pm$15v was chosen because this was most compatible

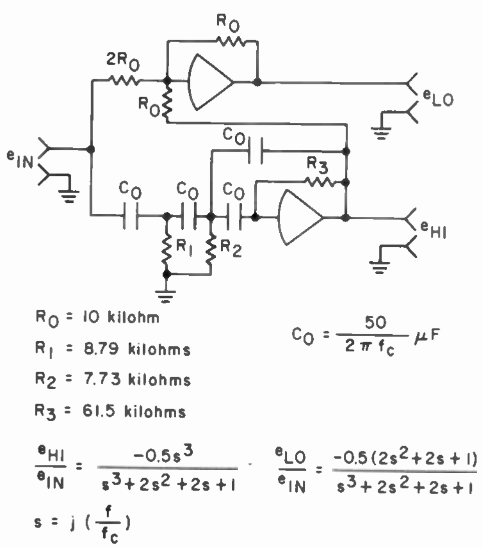

Fig. 2—Active filter network.

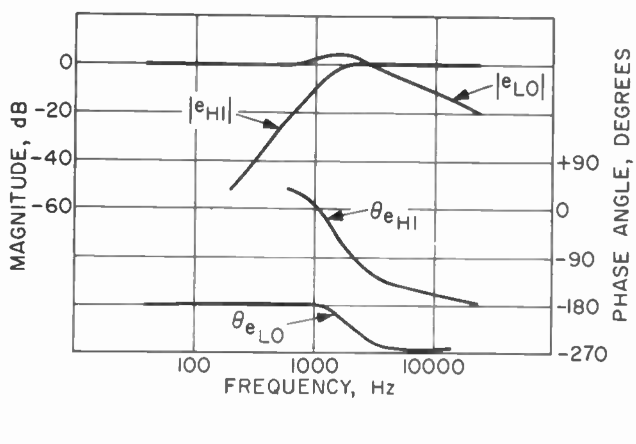

Fig. 3—Measured response of I.C. filter network.

F–Hz	Co-(μF)	F–Hz	Co-(μF)
100.00	0.07958	1258.93	0.00632
125.89	0.06321	1584.89	0.00502
158.49	0.05021	1995.26	0.00399
199.53	0.03988	2511.89	0.00317
251.19	0.03168	3162.28	0.00252
316.23	0.02516	3981.07	0.00200
398.11	0.01999	5011.87	0.00159
501.19	0.01588	6309.57	0.00126
630.96	0.01261	7943.28	0.00100
794.33	0.01002	10000.00	0.00080
1000.00	0.00796		

Table 1—Capacitor values for active filter network.

with the commercially available integrated-circuit power amplifiers and it was intended that both filter and buffer amplifiers could be powered from a common supply.

For this network, the slope of the high-pass characteristic in the stop-band is 18 dB per octave and the phase goes from -180 degrees in the pass-band to 90 degrees in the stop-band, with -45 degrees at the crossover frequency. For the low pass, the stop-band slope is 6 dB per octave and the phase response goes from -180 degrees in the pass-band to -270 degrees in the stop-band with approximately -180 degrees of phase shift at the crossover frequency. Ideally, perhaps, the phase shift in the pass-bands would be zero degrees; however, the operational amplifiers require an additional -180 degrees for stability because of the feedback employed. Note also that a filter could have been synthesized by deriving the low-pass and subtracting to obtain the high-pass. This would also be a constant voltage network and for a third-order low-pass would have an 18 dB per octave slope for the high-pass. The disadvantage to this arrangement is that, because of the greater energy in the low-frequency portion of the spectrum, the tweeter would be forced to handle considerably more power because of the lesser stop-band slope, and this might adversely affect reliability. Furthermore, because of their built-in inductance and reasonably civil behavior above the usual cutoff frequencies, today's low-frequency drivers are more suitable for use with a 6 dB per octave stop-band slope.

Component values were selected from Table IV of Ashley and Kaminsky's paper,[2] which is reproduced here as Table 1, for a crossover frequency of 1584.89, which is suitable for an 8-in. woofer. The values specified by Ashley and Kaminsky are:

R_0 = 10 Kilohm
R_1 = 8.79 Kilohm
R_2 = 7.73 Kilohm
R_3 = 61.5 Kilohm
C_0 = 5.02 Nanofarad

Since these values are not readily available, some change was necessary as shown below.

R_0 = 10 Kilohm
R_1 = 8.79 Kilohm
R_2 = 7.8 Kilohm
R_3 = 60.7 Kilohm
C_0 = 5.0 Nanofarad

This was found to have a negligible effect on the actual response, as demonstrated by the fact that the measured -3 dB point of the high-pass signal (crossover frequency) was 1.55 kHz which is in error by less than 5 percent.

The measured frequency response is as shown in Table 2 and Fig. 3. We note that this is a very flat curve without unexplained bumps and glitches that sometimes plague passive networks of this kind. Also note that the stop-band slopes and phase response are as predicted by myself, and Ashley and Kaminsky.[2]

In short, it is felt that the active filter network is a good performing, easy to realize alternative to the passive networks used in the past. In addition, given the relatively more common component values used as compared with a passive network for a similar function and the low cost of integrated-circuit operational amplifiers (the μA741 is available for less than $1.00), the cost differential should not be excessive. Æ

	e_{hi}		e_{lo}			e_{hi}		e_{lo}	
Freq., -Hz	Mag., dB	Phase, degrees	Mag., dB	Phase degrees	Freq., -Hz	Mag., dB	Phase, degrees	Mag., dB	Phase degrees
40	-50		0	-180.0	1200	-7.2	-12.7	2.9	-184.0
50	-50		0	-180.0	1400	-4.3	-33.4	3.5	-192.7
60	-50		0	-180.0	1700	-2.3	-62.9	3.6	-206.7
80	-50		0	-180.0	2000	-1.0	-81.9	2.8	-217.6
100	-50		0	-180.0	2500	-.4	-104.1	1.4	-229.5
120	-50		0	-180.0	3000	-.2	-118.4	-.1	-238.2
140	-50		0	-180.0	4000	-.1	-135.6	-2.6	-245.5
170	-50		0	-180.0	5000	-.1	-145.2	-4.5	250.0
200	-52		0	-180.0	6000	0	-150.1	-6.0	-251.8
250	-47		0	-180.0	8000	0	-158.3	-8.0	-253.7
300	-42.2		0	-180.0	10000	0	-163.1	-10.6	-255.9
400	-34.8		0.1	-180.0	12000	0	-166.7	-12.3	-258.5
500	-28.9		0.2	-178.8	14000	0	-169.0	-13.8	-261.9
600	-24.3	42.1	0.4	-177.7	17000	0	-171.4	-15.8	
800	-17.0	27.4	1.1	-177.1	20000	0	-173.1	-17.5	
1000	-11.3	6.9	2.0	-178.8	25000	0	-179.4	-20.0	

Table 2—Measured response of I. C. active filter.

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LOUDSPEAKER DESIGN

Hofmann's Iron Law -

a curiously useful way of looking at the low frequency performance of loudspeakers...

HENRY KLOSS*

THERE HAS always been a certain willingness to suspend both disbelief and rationality in the discussion of loudspeakers, and for the most part it doesn't really harm anything. People tend to wind up buying their speaker systems on fairly reasonable, pragmatic grounds, and aren't likely to be disturbed by the adman/salesman/theoretician who holds that it takes a round speaker to yield "round sound" or a speaker system the size and shape of a bass viol to reproduce the sound of one. We do, of course, see some nicely rounded and bass-viol-sized speakers in stores as a result, but the relatively few people who buy them probably have their reasons and can't be considered the worse for it.

Still, there are times when a loudspeaker designer has pangs and longs to sneak a bit more enlightment into the discussion—even if it makes clear, as the following may, that just about anyone might design a good-to-wonderful low-frequency loudspeaker system by following rules that are both few and simple.

So, then, a catharsis for a speaker designer. And an attempt at some new and hopefully useful ways of looking at the low-frequency performance of the kind of speaker system that has such wide acceptance as a high-performance device. That, of course, is the sealed-box, acoustic-suspension system now made (in various adjectival forms) by just about every speaker manufacturer. We won't argue the possibility of a better design somewhere in some better world, but simply proceed with the knowledge of the present design's sublime usefulness in this one.

*Advent, Inc.

One of the delightful things about the sealed-box, acoustic-suspension, single-degree-of-freedom speaker is that it's a quite simple system, with an attendant lack of eccentricity. The parameters that decide its behavior are *there* all the time, and you can vary them for different objectives—as we will be discussing—with known results. You don't wind up with egregious behavior in some performance area as a result of some apparently harmless change, and you don't then have to waste time looking for some "patch" that may itself have some strange effect.

Which means, of course, that low-frequency performance can't be determined by the sheer weight of money or the designer's ingenuity, since the rules stay the same whatever the designer's resources. But while this may disturb those designers who like to think they can buy their way out of a design limitation, or those who think that a particular kind of voice coil or cone material or construction has a certain mystique, it's very nice for all the rest of us. And it does leave the designer free to make some choices, hopefully enlightened, of what to emphasize and what not to emphasize, since total subjective effect or suitability isn't as nicely predictable as are curves and such.

So, while the behavior at low frequencies of a certain sort of speaker isn't the most metaphorically exciting subject for an article, it does give us a chance for a close look at some reasonably interesting things that can be stated both simply and rigorously. Nothing new, really, except perhaps a new window on reality.

What I propose to do is being with a speaker design of known excellence and discuss its basic and completely dependable interrelationships: What happens to performance from various physical changes, what physical changes are needed for a specific performance objective. The assumptions (forgetting about the vital question of your interest in all this) are:

- That we are talking about the sealed or effectively sealed speaker system. (Some latter-day ported systems are essentially sealed boxes that follow the rules we will be discussing.)
- That the speaker derives all damping from its voice coil moving in the field of its magnetic structure and is used with an amplifier of modern high-damping-factor (4 or above) design.
- That the amplifier is not tailored in frequency response to a particular speaker.
- That we aren't making any judgment on how much sound must be produced, but working within known and accepted parameters for average to demanding home use.
- That we aren't after the discovery or definition of one "ideal" loudspeaker, but discussing possibly useful variations within an area of known goodness (or, if you prefer, excellence).

The discussion which follows will be different from the usual presentations in an important way. We shall deal only with those parameters whose manipulations are at the discretion of a designer. By eschewing the inclusion in our statements of such quantities as the density of air, the velocity of sound, the value of 2π, and other constants that are constant for all speakers in this group, we are forbidden to make statements of equality in connecting physical parameters with performance characteristics.

We can, and shall, make perfectly rigorous statements of proportionality which will permit us to *precisely* predict the performance of any new speaker as a function of the change in parameters of a prototype speaker.

If we are told, and we should certainly readily believe, that the weight of a pile of jelly beans is proportional to the number of jelly beans, we should be quite confident that if we multiply the number of jelly beans by 1.2, the total weight will increase by 1.2. Note that we did not have to know how much a jelly bean weighed. If our job is to manipulate the number of jelly beans and then keep track of total weight, we shouldn't concern outselves with those things (constants, i.e., weight of individual jelly beans) over which we have no control. If we are really dedicated to our job of getting at the essential truth, we can even readily accept the fact that these jelly beans are in a fixed size container whose weight does not appreciably disturb the relationship between number and weight of jelly beans over the range in which we are interested (see assumptions above). The whole presentation is directed toward an attempt to make a powerful final statement that connects together those several characteristics which directly affect the value of a loudspeaker to a user.

A good place to begin is the area of greatest comfort to any speaker designer: The frequency range from 800 Hz down to the point below 150 Hz where variations in low frequency curves may begin to be visibly and audibly significant. What is of such comfort about the 150-800 Hz range, as has been stated elsewhere many times, is that it's "flat" by nature (1). Over that frequency range, the speaker's velocity and hence output is controlled by the mass of its moving system. Assuming good design as we are throughout, in this case of the cone, there is ideal piston operation. Cone velocity goes up as frequency goes down, doubling for each halving of frequency (a fact with which Mr. Klipsch apparently likes to frighten small children) to coincide nicely with the realities of decreasing radiation resistance. Output can be calculated precisely at any point in the range as the square of cone velocity times the square of the area times some constants. No trickery is needed to make it come true nor are any special cone materials (a wide variety of thoroughly conventional materials and compositions will do nicely).

But things change as the bass resonant frequency of the system is approached. In a proper closed-box system, output beings to drop at a point somewhere above resonance (we'll be more specific in a moment), drops more at resonance, and begins to roll off fairly sharply (12 dB/octave) somewhere below that as stiffness reactance *halves* the cone velocity for each lower octave. If benign nature seemed to rule in the 150-800 Hz range, the designer takes responsibility now for everything, including (a) the shape of the roll-off, resonance curve, (b) where it begins laterally on the frequency scale, and (c) where the curve *and* the resonably straight line between 150 and 800 Hz show up on the vertical scale of absolute power output.

He is responsible, all right, but the rules are the rules.

The shape of the frequency response curve of *every* speaker of the type we are discussing will inevitably correspond to one of the family of these familiar universal resonance curves. We can construct a graph with explicit labels for x and y axes which completely describes the speaker's performance quantitatively if we know three performance characteristics:

A. *Efficiency*, or the amount of output in the "flat" region for a given power input,
B. *Resonant frequency*, or the actual frequency at point labeled FR on curve, and
C. *Which shape of curve.*

Now, there are four, and only four physical parameters that in turn set those three performance characteristics:

1. A = area of cone,
2. M = mass of moving system, (cone and voice coil largely),
3. Motor = "strength" of the magnet-voice coil motor (2), and
4. V = volume of air enclosed in sealed box.

Let us relate physical parameters to performance characteristics:

A. *Efficiency* $\triangleq A^2/M^2 \times$ Motor;
B. *Resonant frequency*, which we shall call:

$$F_r \triangleq \sqrt{\text{stiffness/mass}}$$

Stiffness here is assumed to be solely due to cone area pressing against a small enclosed volume of air and as such is approximately equal to A^2/V so that:

$$F_r \triangleq \sqrt{A^2/VM}$$

C. *Shape of curve* is actually determined by the "Q" of system at resonant frequency. The relating of Q_{FR} to the physical parameters is a somewhat messy expression, involves all four of those parameters and does not readily permit a feel for the physical situation. We would like now to introduce a different term which relates to shape of curve that makes it much easier to figure out the new performance of a speaker when any one physical parameter is changed. Use of this new term will then lead us to a way to make a powerful and simple statement. We shall also relate this new term to Q_{FR} as we must be able to. They are both, after all, equally legitimate ways to describe the shape of curve.

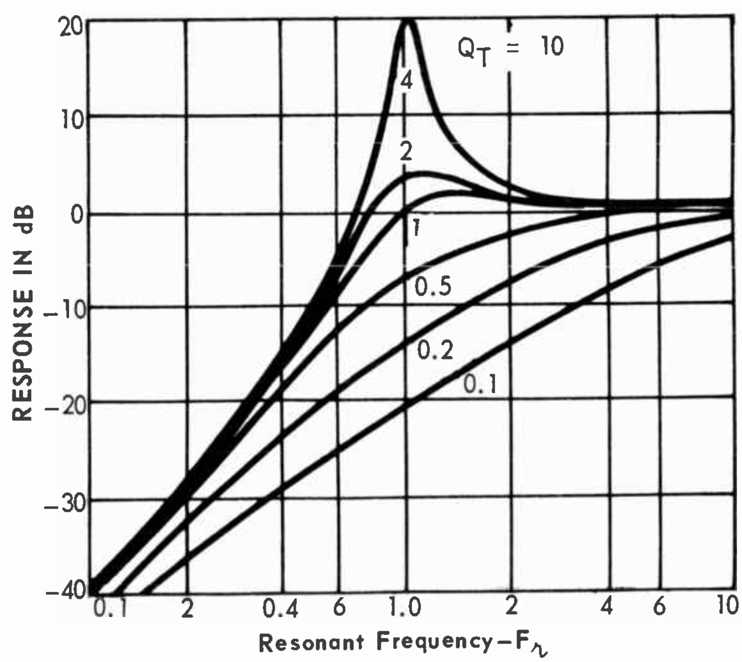

Fig. 1—Frequency response versus Q.

Short Digression

Imagine a loudspeaker with no stiffness at all, that is a resonant frequency at 0 Hz.

Now, let us examine the output as we move down in frequency. We know there is a region of constant output and might expect this to continue to indefinitely low frequencies; we shall certainly never get to the 12 dB octave slope caused by stiffness reactance. Since this loudspeaker must have a motor (some volume of conductor in a magnetic field, here assumed to be fed from a low impedance), there must be some damping force, which, no matter how small, will at some low frequency equal the continually decreasing mass reactance and cause the output to decrease by 3 dB. We shall find it convenient to express the very important relationship between mass reactance (tendency for velocity to increase with lower frequency) and the resistive damping force (tendency for velocity to remain constant) as the frequency at which the two become equal. This we shall call the damping frequency. A stronger motor, i.e., more damping, will cause this frequency to be higher; a heavier moving system, more mass reactanct, will cause this frequency to be lower.

We can see that this quantity is in no way dependent on area of cone or volume of enclosure, but is just a way of describing the relative influence on the velocity of the cone at any frequency of the mass of the moving system and of strength of the motor, and one can readily see that damping frequency which we shall call FD approximately equals motor divided by mass.

To gain familiarity with damping frequency, FD, imagine a speaker with FD at 240 cps and resonant frequency, FR, at 60 cps. If we examine the shape of the frequency response curve going down in frequency from the flat region, we are told that already at 240 cps the damping force is significant compared to the mass reactance in determing cone velocity, and the output is thus below the flat region and shall be even lower by the time we move down to 60 cps. Now take a speaker with FD 15 cps and FR 60 cps. As we move down in frequency from the flat region, we see that when we get to the resonant frequency, the "damping force" is still not a strong contributor to determining cone velocity and, since mass reactance at this frequency (by definition) is cancelled by stiffness reactance, velocity, and hence response, is allowed to rise appreciably. One more example: A speaker with FD = 60 cps, FR = 60 cps. Remembering our definition of damping frequency, this speaker would have been down in response by 3 dB if there were no stiffness at all, i.e., resonant frequency = 0. Because resonant frequency is 60 cps, the mass reactance, which is equal to "damping force," is cancelled by stiffness reactance and the response is allowed to double, i.e., rise 3B to the level of the "flat" region. From this fact you can readily pick out the appropriate curve, namely QFR = 1. (See Fig. 1) In fact, the curve fitting the other two speakers examined can be readily found by making use of the relationship between QFR and FD that

$$Q_{FR} = F_R/F_D.$$

This is just a consequence of the way we have defined FD. Our first speaker is thus seen to have the curve corresponding to QFR = .25 and speaker number 2 has QFR = 4. One might complain that at the beginning we should have just *said* that FD = FR/QFR but that would have denied us the chance to get some physical "feel" for FD and to see why logically it is approximately equal to motor divided by mass.

So our digression has given us a way to express performance characteristic C in a slightly indirect way by specifying FD.

To then find shape of curve we note ratio of resonant frequency to damping frequency which gives us QFR to enable us to assign the proper curve. So for *C* we then write damping frequency approximately equals motor divided by mass.

This relating of physical parameters to performance characteristics makes it quite easy to readily identify all changes in performance when any one of the four physical parameters are varied. One can quickly go through the four examples: 1. Increase area: increase efficiency, increase FR. 2. Increase volume: decrease FR. 3. Increase mass: decrease FR, decrease efficiency, decrease damping frequency (QFR goes up). 4. Increase motor: increase efficiency, increase damping frequency (QFR goes down).

This is quite handy for a speaker designer but the interrelationship of a different set of characteristics has much broader importance. The loudspeaker buyer-listener is not, or should not be concerned with mass of system, area of cone, or strength of motor. None of these individually are separately discernible to a buyer-listener as being proper or improper. I believe we *can* identify three outstanding characteristics that truly determine the value of a speaker to *user* (remembering that we are here concerned solely with low frequency performance). This value, after all, at least here, is the most proper concern. Our intended service here is to show how these value characteristics are rigorously tied together in a very simple way.

The value characteristics are:

1. *Volume of enclosure.* The smaller the better. This strongly affects the utility of the speaker with respect to allowing optimum placement and even more strongly affects price.

2. *Efficiency.* The higher the better. Total loudness for given electrical power.

3. *Low frequency response performance,* which we have shown to be defined by:

   3a. *resonant frequency.* The lower the better.

   3b. *damping frequency.* The lower the better.

   (Assuming we are discussing a properly designed high performance speaker in which the motor, for sake of reasonable efficiency, has been increased beyond the point that frequency response alone would like, i.e., the speaker is overdamped.)

It turns out that these four quantities are closely interdependent. The exact statement of this interdependence turns out to be very pleasing for its simplicity, which is the reward that should be expected for the effort to acquire this new conceptual tool of damping frequency.

Let us again express each of these "user value characteristics" in terms of their dependence on physical parameters:

1. Volume ≙ volume
2. Efficiency ≙ A^2/M^2 x Motor
3a. $F_r \triangleq \sqrt{A^2/MV}$
3b. $F_d \triangleq$ Motor/M

Now just a few lines of old math eighth grade algebra. From 3a, squaring each side we have:

$$F_r^2 \triangleq A^2/MV$$
or
$$A^2 \triangleq F_r^2 MV$$

Let us restate efficiency, substituting for A2 as:

$$\text{Efficiency} \triangleq F_r^2\ MV\ \text{Motor}/M^2$$
$$= F_r^2 V\ \text{Motor}/M$$

But we recognize:

$$\text{Motor}/M \text{ as } F_d$$

So we finally get:

$$\text{Efficiency} \triangleq F_r^2 F_d V$$

If one wants to consider only a given shape of curve, that is a given QFR, we can then express FD as some factor of FR and further simplify our law to:

$$\text{Efficiency} \triangleq F_r^3 V$$

(Continued on page 56)

These then are the quantitative expressions of what I have come to call Hofmann's Iron Law[3] *prescribing the amount and direction of change that* must *occur in one or two or three of the remaining terms when any one is changed.*

Note that this law does not say what happens to any one term as a given term is changed, that is, for instance we are not told how efficiency changes if the volume changes. (In fact, it doesn't change at all.)

Once it is decided to vary any term in this statement, it is up to the designer to rearrange physical parameters both to accomplish this change and properly apportion the necessarily resulting change in the other terms to make the most acceptable "new" speaker.

This iron law which shows difficult and frustrating constraints facing an engineer (but apparently not every advertising department) can also console one that the "improvement" one can make over an already properly designed speaker must be nil, independent of his resources or intelligence. A very constructive use can be made of this law by noting not only what it *requires* but what it may *allow*. Physical laws are not inherently malevolent, after all. For instance, we might observe that one could start with a loudspeaker of truly distinguished low frequency performance and keeping F_R and F_D the same have exactly that same shape of curve in one half the volume at 3 dB less efficiency. Arguing the possible value of such a special speaker is outside the scope of this article; our only intent here is to prove that it is possible and even indicate how the physical parameters should be juggled to achieve the result. Æ

1. Technical articles back to Rice and Kellogg's of almost 50 years ago have described this natural occurence of ideal flat behavior. More recently AUDIO covered the subject well in the March 1970 loudspeaker issue. Our treatise assumes a vague-to-working familiarity with the content of such tracts. We are just offering a statement of consequences of the facts which have been well reported.

2. This parameter, which occurs in the expression for efficiency (establishing the force delivered to the moving system as a function of electrical power input) is the same parameter that determines the resistance offered to the moving system by a shorted coil in a magnetic field which, although it is an impedance (real), we are calling "damping force." The expression for strength of motor in terms of physical parameters is approximately magnetic field times volume of conductor. Since those terms always occur together in describing performance characteristics, we shall carry them along, describing them solely as "motor." The designer can then decide how he apportions values between magnetic field and volume of conductor to get the required motor and then decide how he subdivides this volume of conductor to get his desired impedance level. To drag along length of conductor, etc., through all which follows would really obscure the picture.

3. Dr. J. Anton Hofmann first introduced the term "damping frequency" about 16 years ago when the author was struggling with the design of the first of this then-new type of loudspeaker, the AR-1. The use of the term as a manipulative device and, more importantly, the expression of the interrelationships of the "value in use" characteristics that the use of this term revealed were a powerful tool that permitted the designer easily to systematize the design of low frequency loudspeakers. Dr. Hofmann later lent his initial to another loudspeaker company and subsequently has become treasurer and chief-enforcer-of-rigor of Advent Corporation.

The Why and How of Horn Loudspeakers

Victor Brociner*

Horn loudspeakers are used whenever high efficiency is required and when special control is desired over directional characteristics. In a well-designed horn speaker, distortion is lower than in a direct radiator, at least towards the low-frequency limit of its range. Bass horns are capable of producing high sound pressure levels with a small cone excursion; because of this, both nonlinear and Doppler distortions are lower than in direct-radiator woofers. Direct-radiator speakers become more directional as the frequency increases. Horns can be designed to provide good directional patterns up to the highest frequencies.

The theory of horn speakers to be found in textbooks is highly mathematical, difficult to understand, and very abstract. It does not provide a good intuitive understanding of horn speaker operation. Popular books, on the other hand, tend to present ostensibly simple explanations that are really not much more comprehensive. What is mean by saying that a horn is an impedance transformer? How does a horn "load" a driver unit? When a speaker is loaded by a horn, doesn't the increased load reduce the diaphragm excursion? If so, why does the acoustic output increase? This article is an attempt to explain horn speaker operation in terms that provide both an intuitive and a quantitative understanding, without using advanced mathematics. It will be assumed that the reader has some acquaintance with electrical analog circuits for mechanical and acoustical systems, and with concepts such as radiation impedance, motional impedance, mechanical compliance, and the like. It is freely confessed that, in the course of the development, some shameless simplifications will be made. The following is certainly not represented to be a design manual.

Direct radiator speakers have low efficiency because there is a bad mismatch between the impedance of the mechanical power source and the air into which it radiates sound, except at resonance, where *electrical* mismatching reduces the output. Fig. 1 is an analog circuit for a direct-radiator speaker in its middle frequency range in which losses are neglected. In this circuit, the acoustic output is the power produced in the radiation resistance R_{MA} by the velocity v:

$$P_A = V^2 R_{MA} \qquad (1)$$

Over the flat section of the frequency response curve shown in Fig. 2, the motion of the cone per unit of applied force is determined almost entirely by the mass of the moving system:

$$V = \frac{F}{2\pi f m} \qquad (2)$$

where v = cone velocity in meters per second
F = force applied by the voice coil, in Newtons
f = frequency, in Hertz
m = mass of the moving system (coil and cone) in kilograms

Equation (2) shows that cone velocity is inversely proportional to frequency. Figure 2 depicts this variation. The dashed curve shows how v^2 varies with frequency. The radiation resistance, R_{MA}, is a function of the square of the frequency, as shown by the dot-dash line. Now, referring to equation (1), the acoustic power output is derived as follows:

$$V \propto \frac{1}{f}$$

$$V^2 \propto \frac{1}{f^2}$$

but $$R_{MA} \propto f^2$$

so $$\rho_A = V^2 R_{MA} \propto \frac{1}{f^2} \cdot f^2 = \text{constant} \qquad (3)$$

This is seen in the power-frequency response curve of Fig. 2. Suppose the voice coil mass plus cone and air load mass equals .03 kg. At 400 Hertz, where the frequency response is flat, the reactance of the mass is

$X_m = 2\pi fm = 6.28 \times 400 \times .03 = 75$ mechanical mks ohms

This is the effective impedance of the generator. The load,

$R_{MA} = 1.57\omega^2 a^4 \rho o/c$*

where $\omega = 2\pi f$
a = radius of piston in meters
o = air density in kg/m^3
c = velocity of sound in m/sec.

Simplifying: $R_{MA} = .0132\ d^4 f^2$
where d = piston diameter in meters (5)
for a 10-inch cone, d = .24 meter approximately
At f = 400 Hertz

$R_{MA} = .0132 \times (.25)^4 \times 400^2$
$= 8.25$ mechanical mks ohms

So a 75-ohm generator feeds an 8.25-ohm load. This indicates the degree of mismatch in a typical case of a direct radiator speaker.

Matching can be improved by reducing the generator impedance or increasing the load impedance. The reactive generator impedance can be cancelled out by resonating its mass with a compliance. This is what happens at the resonance frequency of a direct radiator. At and near this frequency, the impedance match is good, and the mechano-acoustic efficiency is high. However, the low mechanical impedance that occurs at resonance is reflected into the electrical circuit as a high impedance, as a result of which the speaker cannot absorb much electrical power from the amplifier, that is, if the speaker is well damped. If not, there is bump in the response curve at resonance. This is not the kind of efficiency we want.

The reactive generator impedance can also be reduced by reducing the mass, m. This is done in tweeters by using small, aluminum voice coils and very light cones. In wide-range speakers or woofers, there is a limitation to mass reduction because the cone must be sufficiently strong so as not to buckle or break up into modes. Another problem is that reducing the mass raises the frequency of resonance, below which the frequency response falls rapidly. It also reduces the Q of the moving system, since

$$Q = \frac{2\pi fm}{R_m} \qquad (6)$$

where R_m is the damping resistance. An over-damped speaker has a frequency response that falls off as the frequency de-

*Vice President Engineering, H. H. Scott, Inc.

*Leo L. Beranek - Acoustics 1954 P. 124 Table 5.1

creases. All of these factors are undesirable consequences of excessive lowering of mass.

This leaves the factor R_{MA}—the radiation resistance. Since R_{MA} increases proportionally to the area of the speaker, it is a simple matter to make it larger. But increasing the size of the cone raises its mass; rather rapidly, in fact, because the thickness has to be increased as well to maintain adequate stiffness. This works to counteract the increased efficiency obtained by the larger cone. Also, as examination of the frequency response curve of Fig. 2 discloses, the point at which the high-frequency response falls off is lower the greater the cone diameter. For a theoretically perfect piston, the drop in response begins at f = 8600/d where d is the piston diameter in inches. Thus, a 10-inch piston begins to drop in response at 860 Hertz, a 15-inch diameter results in a 573-Hertz transition point, and so on ----- downwards.

Increasing Radiation Resistance

If we could somehow manage to persuade a cone to produce plane waves, the radiation resistance would be 407 mechanical mks ohms per square meter of radiating area. A 10-inch cone, with an area of .05 square meter, would then work into 407 × .05 or 24 ohms which is certainly more favorable than the 8.25 ohms calculated in the example given above. We know that sound that is not too high in frequency is propagated as a plane wave along an infinitely long pipe. We could presumably get better efficiency by coupling the speaker to such a pipe. However, we might not care to wait around until the sound emerged from the pipe! What about a very long, but not infinite, pipe? The load on the speaker would be favorable, but now, at the mouth of the pipe we face much the same situation as we had with the speaker radiating directly.

The only way around this problem is somehow to attain a larger area at the point of transfer of sound from the pipe to the surrounding air. Fig. 3 shows a way of doing this. The expanding passage employed is called a horn.

To determine exactly how a horn works even with simplifying assumptions, requires abstruse mathematical treatment. However, there is a simplified approach which, although anything but rigorous, can convey a rather good idea of what goes on in a horn. Let us consider the propagation of a plane sound wave through a small pipe that feeds a larger one as shown in Fig. 4. Assume that both pipes are infinite in length. The acoustic resistance of the small pipe is 407 S_1 mechanical mks ohms, S, being its cross-sectional area in square meters. The large pipe has an acoustic resistance of $407S_2$. The resistances are not matched at the junction, so some reflection takes place, and not all of the power is transmitted from one pipe to the other.

The ratio of power transmitted is

$$r_p = \frac{4r_s}{(1 + r_s)^2} \qquad (7)$$

where $$r_s = \frac{S_2}{S_1}$$

It is interesting to note that S_2/S_1 can be appreciably larger than 1.0 without a great deal of power loss. For example, a 1.5 to 1.0 ratio results in 96% transmission of the incident power. This suggests that it might be feasible to transfer power without excessive losses to a larger area by means of successive small steps of expansion,* as in Fig. 5. Calculations on increasing numbers of steps of expansion for a given ratio of S_2 to S_1 reveals a definite trend that can be seen in the following:

*It is not correct to couple steps together like this because there is an impedance mismatch, but in this case the end justifies the means.

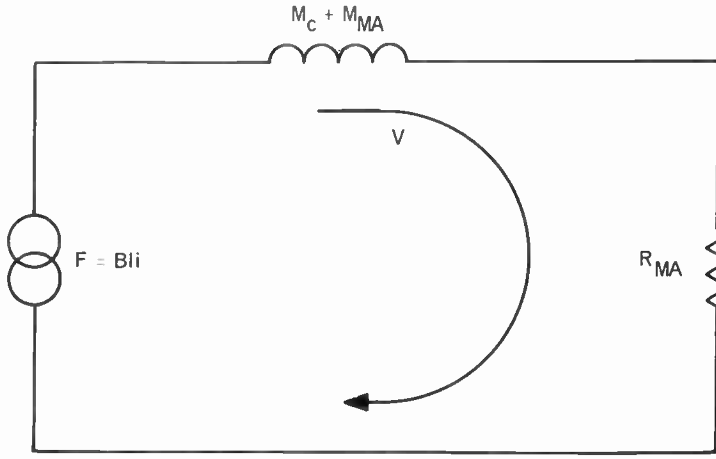

Fig. 1—Mechanical analog circuit of direct radiator loudspeaker with a constant current drive. Frictional losses are assumed as zero. F = force exerted by voice coil in Newtons; l = length of voice coil conductor in meters; i = current in voice coil in amperes; m_C = mass of voice coil and cone in kilograms; m_{MA} = equivalent mass of air load in kilograms, and R_{MA} = radiation resistance in mks mechanical ohms.

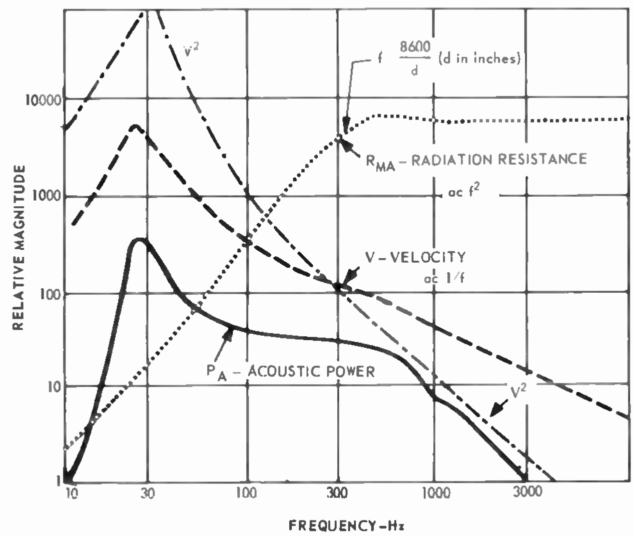

Fig. 2—Calculated performance of piston as direct radiator.

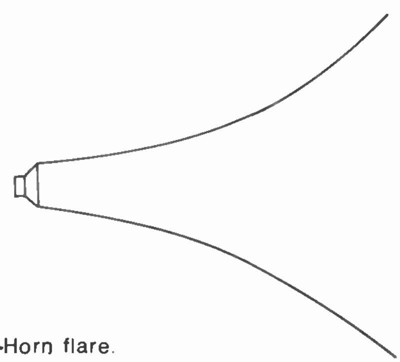

Fig. 3—Horn flare.

Ratio of areas per step	Number of Steps	r_p per step	Total r_p
2.00	1	.89	.89
1.41	2	.97	.94
1.26	3	.99	.97
1.19	4	.992	.98

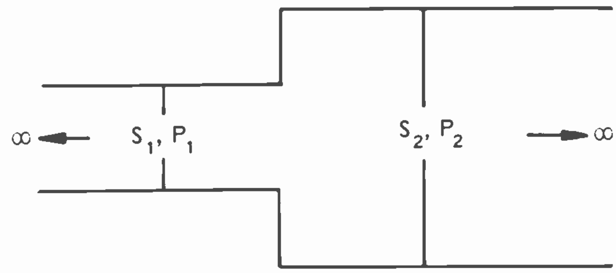

Fig. 4—Acoustic power transmission through stepped pipe. S_1. S_2 = cross-sectional areas; P_1, P_2 = power transmitted, and $r_P = P_2/P_1$.

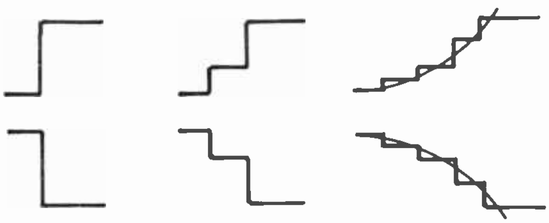

Fig. 5—Limit of successively larger numbers of steps of expansion of a pipe for identical initial and final diameters.

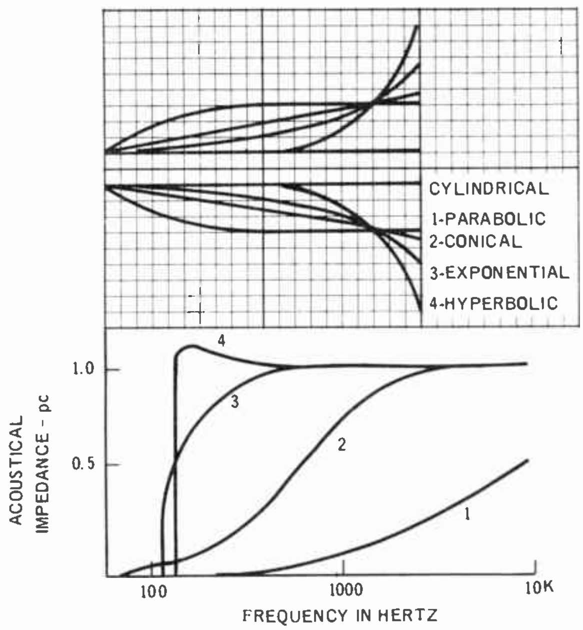

Fig. 6—Expansion rates of various types of horns. S = area at distance X from throat; S_T = throat area. Parabolic: $S = S_T(1 + X/X_o)$; Conical: $S = S_T(1 + X/X_o)^2$; Exponential: $S = S_T e^{MX}$; Bessel: $S = S_T(X + X_o)^M$; Catenoidal: $S = S_T(e^{MX} + e^{-MX})$; Hyperbolic: $S = S_T(\cos H\, X/X_o + T \sin X/X_o)^2$ T is less than 1.

In all four cases, the ratio of output area to input area is 2.00 and it is evident that increasing the number of steps raises the efficiency of transmission. The trend appears to indicate that the limit of a continuous curve is 100% transmission. Now, if the far end of the pipe can be made large enough so that it produces plane waves, just as plane waves would be radiated by a large piston, the finite pipe behaves like an infinite pipe. There is no problem at the far end, power is transmitted 100% along the pipe (except for frictional losses) and the desired increased load is obtained on the diaphragm of the driving speaker. The device we have "invented" is a horn; the end near the speaker is called the *throat* and the far end the *mouth*.

Horns: Infinite and Finite

The shape of a horn is expressed by a mathematical formula indicating the ratio of the horn cross-sectional area at a given distance, x, from the horn throat, to the area of the horn mouth. The most widely used horn is is *exponential*:

$$\frac{S_2}{S_1} = e^{mx} \qquad (8)$$

where S_2 = mouth area
S_1 = throat area
m = flare constant
x = distance from throat

In this type of horn, each increment of distance results in a constant ratio of areas:

If $S_2 = S_1 e^{mx}$

then, for an increase of x in distance

$$S_2 = S_1 e^{m(x+\Delta x)} = S_1 (e^{mx} \cdot e^{m\Delta x})$$
$$= S_1 e^{mx} \cdot e^{m\Delta x}$$
$$S_2/S_1 = e^{m\Delta x} \text{ is a constant}$$

This is the type of horn we developed as a limit of the stepped pipe as the number of steps per unit distance increased indefinitely. Other useful horn shapes are Bessel, catenoidal, conical, hyperbolic, and parabolic. These are all shown in Fig. 6. The conical horn is the oldest form of artifical horn, seen in megaphones. The oldest natural form is approximately paraboloidal, consisting of a pair of cupped hands held before the mouth. Paraboloidal horns are used, in segments of different flare rates, to approximate horns of other expansion laws, in low-frequency horns. Many folded bass horns, which are usually made of wood, use a succession of short sections with a pair of parallel sides, the other pair expanding linearly. The area of such a horn section follows a parabolic law of expansion.

The hyperbolic and Bessel horns are families of horns, the exact curve depending on the values of certain parameters. The hyperbolic horn becomes exponential when T = 1, and catenoidal when T = 0. The Bessel horn becomes conical when m = 1.

Fig. 7, shows a comparison between the radiation resistance of a loudspeaker radiating directly into the air and the same unit working into an exponential horn. There is a large increase in radiation resistance at frequencies below the knee of the curve for the direct radiator, and none at all for high frequencies. The lack of improvement at the top end is not surprising when one realizes that a direct radiator produces virtually plane waves at high frequencies. These can be visualized as emerging as a beam from the speaker without being affected by the horn walls.

All of the curves go to very low values of radiation resistance as the frequency is decreased below a certain point. In an exponential horn, for example, the radiation resistance becomes zero when:

$$f_c = mc/4\pi \qquad (9)$$

where f_c = cut-off frequency
c = speed of sound
m = flaring constant.

The radiation resistance does not actually go down to zero, but retains a small finite value, not given by the usual horn formula because of the simplifying assumptions adopted in its derivation. At $\sqrt{2}\, f_c$, R_{MA} is $1/\sqrt{2}$ of its ultimate value. A horn driven by a piston whose velocity is independent of frequency has a sound pressure level that is down 3 dB at this frequency.

A convenient, if approximate concept, is that below the cut-off frequency the air in the horn does not propagate as a travelling wave but simply oscillates in the horn so that the pressures in all planes at right angles to the horn axis are in phase. Under these conditions the air acts like a moving mass.

As a matter of fact, the air in the horn does not move exactly in phase with the driving pressure even in its working range; in other words, the acoustic impedance looking into the horn throat is not a pure resistance. The formula for the throat reactance is:

$$X_{MA} = \frac{\rho c^2 m}{S_T 4 \pi f} \qquad (10)$$

in which, be it noted, the frequency term is in the denominator. The reactance is positive, which leads one to conceive of it as a mass reactance, but it *decreases* as the frequency increases, as does a capacitance. Actually, the reactance is that of a negative capacitance. It acts to decrease the velocity for a given applied force, reducing acoustic output.

There is another limitation at low-frequencies as well, which is not shown by the curves of radiation resistance (Fig. 6) because these are based on horns of infinite length. It may be recalled that in developing the horn via the stepped pipe, it was stated that the mouth size had to be great enough for the assumption that it radiated plane waves.

Adequate "mouth loading" is generally considered to be provided when the circumference of the horn is equal to a wavelength, or $\pi d = \lambda$. This is somewhat easier to deal with in terms of frequency: $f = c/\lambda$ where c is the speed of sound. The resulting expression is:

$$d = \frac{c}{\pi f}$$

For d in inches, d = 4300/f. This is exactly one ocatve below the frequency at which the radiation resistance curve of a direct radiator has its knee. (See Fig. 2)

The formula provides some discouraging information. For example, a horn that performs well down to 43 Hertz has to be over 8 feet in diameter! Fortunately one can live with considerably less than perfect mouth loading. Also, the use of room walls and floor as reflectors increases the effective area. Bass horns designed to fit into the corner of a room work very well with equivalent mouth diameters not much greater than 2 feet.

If we now consider the frequency response of the simple horn speaker in the midfrequency range we find that we do not obtain a flat frequency response.

At 400 Hertz we found the mass reactance to be 75 mks mechanical ohms; by means of the horn the radiation resistance load was increased to 24 ohms, but this still leaves the mass largely in control of the veolcity (See Fig. 1). Meanwhile, the radiation resistance has become *independent* of the frequency (See Fig. 7). Since the velocity v still decreases with increasing frequency and R_{MA} remains constant, the formula $P_A = v^2 R_{MA}$ tells us that the response falls toward the high-frequency end. If R_{MA} is frequency-independent, v must be constant as well. This condition can be approached if R_{MA} is much greater than the mass reactance.

R_{MA} can be increased still more if the speaker is coupled to the horn by means of a stepped-down pipe as in Fig. 8. Now the speaker looks into a mechanical resistance

$$R_S = \frac{S_D^2}{S_T} \rho c \qquad (11)$$

where S_D = the diaphragm area
and S_T = the throat area
If $S_D/S_T = 4$; $R_S = 4 \times 24 \times 96$ mks mechanical ohms which approaches the condition we want of resistance-determined cone velocity. With true resistance control, the response is flat in the mid-frequency region. The variation with frequency of the response - determining parameters as well as the response curve itself, are shown in Fig. 9a, and the analog circuit in 9b.

As the frequency increases, the mass reactance begins to predominate in determining the velocity, which decreases. The power output drops at a rate of 6 dB per octave. Eventually, the compliance of the front air chamber C_c begins to act like a shunt across the radiation resistance R_{MA}, and response falls rapidly. If the low-pass filter formed by m_S and C is proportioned so that it is suitably mis-matched, the response in the region immediately below A can be elevated as shown by the dot-dash line with response falling above this point at a rate of 12 dB per octave. See Fig. 9c. At low frequencies the response falls because X_{MA}, as has been pointed out, is a negative capacitance, whose reactance increases, decreasing the diaphragm velocity. However, if the speaker resonance is placed near the horn cut-off frequency, the speaker compliance can be made to cancel out the negative compliance of the horn, removing this restriction and increasing the bass response. In high-compliance woofers, the required lower value of compliance can be obtained by enclosing the rear of the speaker in a very tight box.

Efficiency

We have found that adding a horn to speaker unit improves the impedance matching and increases the power output for a given driving force of the voice coil. What about the efficiency? Let us simplify the problem by considering the frequency range over which the cone motion is resistance-controlled, and neglect losses due to friction. The efficiency under these idealized conditions is called the initial efficiency.

The force exerted by the voice coil:

$$F = Bli \qquad (12)$$

where F is in Newtons
B = magnetic flux density in Webers/square meter.
l = length of voice coil conductor in meters

The cone velocity

$$V = F/R_{MA} \qquad (13)$$

where v is in meters/second
R_{MA} = radiation resistance in mks mechanical ohms.
Substituting from (12) to (13),

$$V = \frac{B \ell i}{R_{MA}}$$

Using the formula for acoustic power

$$P_A = V^2 R_{MA} = \frac{B^2 \ell^2 i}{R_{MA}} \text{ acoustic watts} \qquad (14)$$

The input power consists of the electrical power dissipated in the voice coil, plus that corresponding to the acoustic power.

$$P_E = i^2 R_c + \frac{B^2 \ell^2 i}{R_{MA}}$$

$$= i^2 \frac{R_c R_{MA} + B^2 \ell^2}{R_{MA}} \qquad (15)$$

Efficiency

$$\eta = \frac{P_A}{P_E} = \frac{B^2 \ell^2}{B^2 \ell^2 + R_c R_{MA}} \qquad (16)$$

For a 12-inch speaker with $B^2l^2 = 64$ and $R_c = 6$ ohms, and $R_{MA} = 96$ ohms as in the horn previously analyzed:

$$\eta = \frac{64}{64 + 6 \times 96} = 10\%$$

This is not a startlingly high efficiency, but it is a great deal better than the typical 1% or lower efficiency of a typical high-compliance woofer.

How can efficiency be increased? Equation (16) tells us:

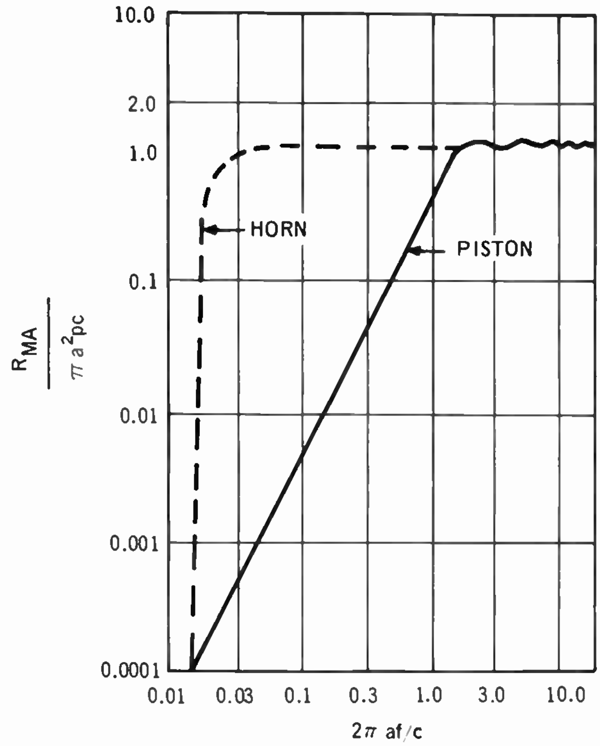

Fig. 7—Radiation resistance of exponential horn versus piston in infinite baffle.

Increase the flux density B, the length of voice coil conductor, reduce the voice coil resistance R_c, or reduce the radiation resistance R_{MA}. The first three items are somewhat interdependent. If l is increased, R_c is increased, opposing the increased efficiency, unless the wire diameter is increased. If this is done, the air gap must be made greater to accommodate the thicker voice coil, reducing B. Also, the moving mass increases, and its greater reactance opposes the condition of resistance control at a lower frequency, reducing the high-frequency response. It can be seen that a great deal of juggling must be done to obtain the desired performance.

Now for the last factor: R_{MA}. To increase efficiency, this must be decreased. But the whole point of horn loading was to *increase* R_{MA}. There seems to be a contradiction here. It is the result of the simplifying assumptions. For the direct radiator, we assumed mass control. For this condition, R_{MA} should be as great as possible. The horn efficiency equation was based on resistance control. Here, R_{MA} should be as small as possible. But if R_{MA} is made small compared to the mass reactance, the condition of resistance control no longer applies. One would expect an optimum point somewhere in between.

If an intermediate condition is assumed between mass control and resistance control, both elements must be taken into account in the expression for the velocity which now becomes

$$v = \frac{F}{\sqrt{R_{MA}^2 + X_M^2}} = \frac{B\ell i}{\sqrt{R_{MA}^2 + X_M^2}} \qquad (17)$$

and

$$P_A = v^2 R_{MA} = \frac{B^2 \ell^2 i^2 R_{MA}}{R_{MA}^2 + X_M^2} \qquad (18)$$

Now the input power is the sum of the output power and the power dissipated in the voice coil

$$P_E = i^2 R_c + \frac{B^2 \ell^2 i^2 R_{MA}}{R_{MA}^2 + X_M^2} \qquad (19)$$

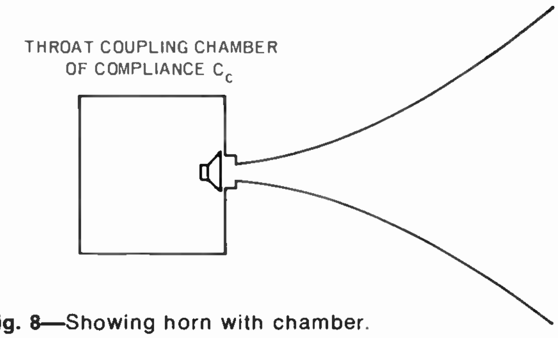

Fig. 8—Showing horn with chamber.

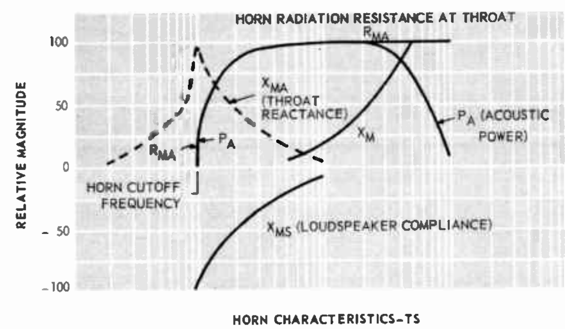

Fig. 9A—Horn characteristics.

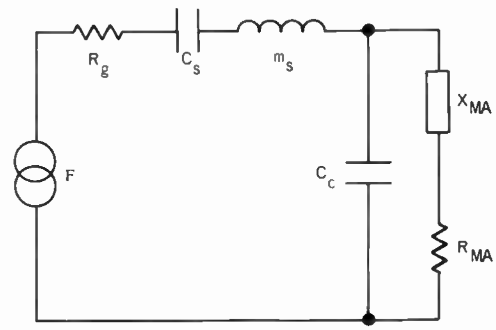

Fig. 9B—Horn loudspeaker analog.

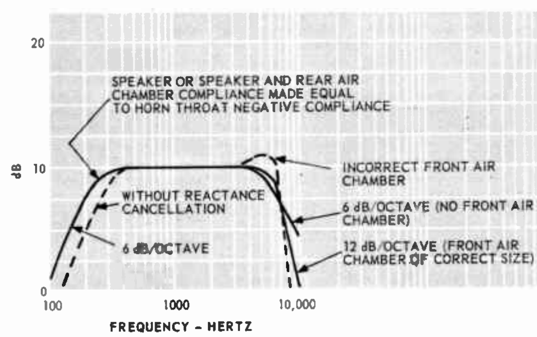

Fig. 9C—Frequency response of horn speaker.

and

$$\eta = \frac{P_A}{P_E} = \frac{B^2 \ell^2 R_{MA}}{R_c (X_M^2 + R_{MA}^2) + B^2 \ell^2 R_{MA}}$$

$$\eta = \frac{B^2 \ell^2}{B^2 \ell^2 + \dfrac{R_c (X_M^2 + R_{MA}^2)}{R_{MA}}} \qquad (20)$$

In this expression, η is maximum when

$$\frac{X_M^2 + R_{MA}^2}{R_{MA}}$$

is a minimum as R_{MA} varies.

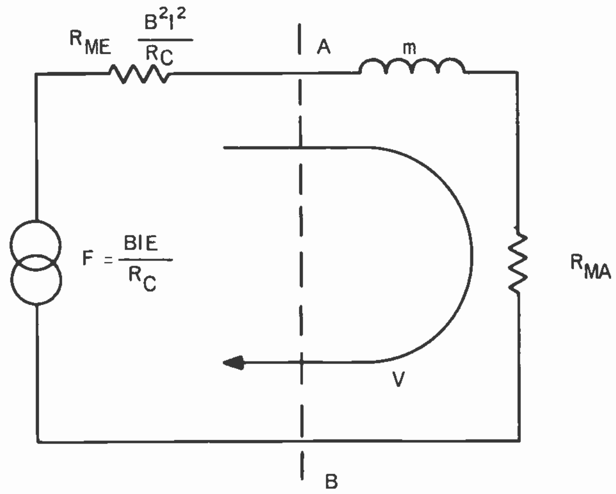

Fig. 10—Analog circuit of horn loudspeaker operated from constant-voltage source. The horn throat reactance is assumed to be cancelled out by the speaker compliance.

This occurs when $R_{MA} = X_M$. Then the maximum possible value of efficiency.

$$\eta_{MAX} = \frac{B^2 \ell^2}{B^2 \ell^2 + 2 R_c X_M}$$

where $X_M = 2\pi f m$ (21)

Since X_M varies with frequency we can select f to produce a maximum anywhere we please as long as $X_M = R_{MA}$ at some point on the flat part of the curve of R_{MA} vs. f.

Assume that this point is 50 Hz. Then for a .03 Kg moving system $X_M = 2\pi \times 50 \times .03 = 9.9$ mks mechanical ohms. If this is matched by making $R_{MA} = 9.9$ ohms also, then for the speaker previously considered as a direct radiator:

$$\eta = \frac{64}{64 + 2 \times 6 \times 9.9} = 41\%$$

For the same system at 500 Hz, $X_M = 2\pi \times 500 \times .03 = 99$ mks mechanical ohms. However, R_{MA} is fixed at 9.9 ohms. Now

$$\eta = \frac{64}{64 + \dfrac{6(99^2 + 9.9^2)}{9.9}} = 1.1\%$$

The high efficiency at 50 Hz has been obtained at the expense of a response that drops rapidly as frequency increases. This is an illustration of the concept of gain-bandwidth product. As the required bandwidth increases the efficiency goes down.

There is still another method of obtaining high efficiency that can be derived from equation (21). This is to reduce X_M by decreasing the moving mass. One way to accomplish this is to use a smaller cone. This is feasible because high values of R_{MA} can be obtained by means of the horn rather than through the use of a large radiating area.

Constant-Voltage Operation

In all the previous calculations it was assumed that the loudspeaker was operated from a constant-current source. This assumption brings about a considerable simplification in the formulas and provides a valid expression for efficiency. However, it does not provide a correct expression for the frequency response of the loudspeaker as actually used: From an essentially constant-voltage source. The reason is that the electric impedance varies with frequency. Because of this, the input power also varies with frequency.

The mechanical analog circuit of the loudspeaker must be modified from that of Fig. 1, when a constant-voltage amplifier is used. The revised circuit is shown in Fig. 10.

M is the mass of the moving system and R_{MA} the radiation resistance in mechanical units as before. The series resistance represents the effect of the voice coil resistance. Its value is derived as follows: Think of a loudspeaker whose voice coil motion is blocked by some means. If $v = 0$, there must be an open circuit in the loop shown in the figure. This corresponds to the manner in which we would measure the open-circuit voltage in an electric circuit. On the electrical side, there is no counter-e.m.f. generated since there is no motion of the voice coil. The only current-determining element in the circuit is R_C, the voice coil resistance. So $i = E/R_C$ where E is the amplifier output voltage. Substituting in the expression for

$$F = Bli \qquad F = \frac{BlE}{R_C}$$

Now consider what happens if the loudspeaker is operated into a vacuum and there is no friction. Nothing opposes the motion of the cone. This corresponds to a short-circuit across A-B. In the mechanical circuit $v = \dfrac{F}{R_{FM}} = \dfrac{BlE}{R_C R_{FM}}$

The velocity must be just enough to generate a counter-emf that equals the amplifier voltage. Then $Blu = E$ and substituting from the previous expression

$$B \ell v = \frac{B^2 \ell^2 E}{R_c R_{ME}} = E$$

The last two terms result in

$$R_{ME} = \frac{B^2 \ell^2}{R_c}$$

From Fig. 10, velocity

$$v = \frac{F}{R_c \sqrt{\left(R_{MA} + \dfrac{B^2 \ell^2}{R_c}\right)^2 + \omega^2 M^2}} \qquad (22)$$

Substituting $F = B \ell E / R_c$

$$v = \frac{B \ell E}{R_c \sqrt{\left(R_{MA} + \dfrac{B^2 \ell^2}{R_c}\right)^2 + \omega^2 M^2}}$$

We can now find a means of calculating the frequency response of the loudspeaker. The relative response is plotted as sound pressure vs. frequency. Since sound pressure is proportional to the square root of the power, it is convenient to think in terms of the ratio

$$\sqrt{P_A} / E$$

which corresponds to sound pressure per volt applied to the loudspeaker terminals. From the relation

$P_A = V^2 R_{MA}$ we can write

$$\frac{\sqrt{P_A}}{ED} = \frac{v \sqrt{R_{MA}}}{E}$$

Substituting the equation (22)

$$\frac{\sqrt{P_A}}{E} = \frac{B \ell \sqrt{R_{MA}}}{R_c \sqrt{\left(R_{MA} + \dfrac{B^2 \ell^2}{R_c}\right)^2 + \omega^2 M^2}} \qquad (23)$$

Since ωm increases with frequency, the response begins to drop above the frequency where ωm becomes comparable in magnitude to

$$\left(R_{MA} + \frac{B^2 \ell^2}{R_c}\right)$$

Using the constants for the horn speaker whose efficiency was previously derived for 50 Hz and 500 Hz.

At 50 Hz.

$$\frac{\sqrt{\rho_A}}{E} = \frac{8\sqrt{9.9}}{6\sqrt{\left(9.9^2 + \frac{8^2}{6}\right)^2 + (6.28 \times 50 \times .03)^2}}$$

$$= 0.42$$

At 500 Hz.

$$\frac{\sqrt{P_A}}{E} = \frac{8\sqrt{9.9}}{6\sqrt{\left(9.9^2 + \frac{8^2}{6}\right)^2 + (6.28 \times 500 \times .03)^2}}$$

$$= 0.30$$

This is a drop in response of 2.5 dB. The previous calculation for constant current operation showed a drop of over 16 dB. Substantially better frequency response can be obtained with a constant-voltage source because the electric impedance of the speaker decreases at high frequencies and permits more current to flow.

Midrange and High-Frequency Horn Speakers

To this point we have considered only the case where a horn is added to a direct-radiator loudspeaker. Where both extreme low-frequency range and power capability are not needed, horn loudspeakers can be made more efficient by using small, light, rigid diaphragms. Small diaphragms reduce the value of R_{MA}, increasing the efficiency; the moving mass is also lowered, extending the high-frequency response. The smaller driver unit is capable of handling large amounts of power without excessive diaphragm excursion because of the greater load into which it works. The expression for the power output of a loudspeaker indicates that, for a given amount of acoustic power at a given frequency, the required rms velocity is inversely proportional to the square root of the mechanical radiation resistance imposed by the acoustic load. The required excursion is proportional to the velocity. A 10-inch come as a direct radiator, with an area of .05 m^2, works into 8.25 ohms at 50 Hertz. Horn-loaded to 96 ohms, its excursion for a given acoustic output would be

$$\sqrt{\frac{8.25}{96}}$$

or about 0.3 times as great as before. Alternatively, the cone size could be reduced and the radiation load maintained by decreasing the area of the horn throat, which would result in a longer horn.

The excursion varies inversely as the square of the frequency; consequently this requirement becomes rather unimportant except in the bass range. If a direct radiator requires a 3/8-inch peak excursion at 50 Hz, this would be decreased to perhaps 1/8-inch by horn loading. At 500 Hz the excursion would be only 0.0013". Since it is easy to obtain appreciably greater excursions than this in small-diaphragm drivers, mid-range speakers for horn loading can be made quite small. A 3-inch diameter diaphragm is typical in such an application. Diaphragms of this type are frequently dome-shaped and made of aluminum alloy or formed plastic. Consequently they are extremely light. Tweeters use even smaller diaphragms and lighter voice coils.

AN EXAMPLE of horn speaker performance is worked out below. The horn unit is designed to operate in the frequency range above 500 Hz as part of a powerful two-way theater speaker system.

R_c = 12 ohms

B = 1.9 webers/meter^2

l = 1.75 meter

$B^2 l^2$ = 22 webers^2/meter^4

S_T = 3.14 x 10^{-4} meter^2

S_D = 28.3 x 10^{-4} meter^2

S_D/S_T = 9

R_{MA} = 9 x 28.3 x 10^{-4} x 407
= 10.4 mks mechanical ohms.

The initial efficiency at frequencies for which the moving mass has negligible effect:

$$\eta = \frac{22}{22 + 12 \times 10.4} = 16\%$$

If the mass is .002 Kg, at 500 Hz: $2\pi f m$ = 6 mks mechanical ohms and at 5 kHz: $2\pi f m$ = 60 mks mechanical ohms

For constant-voltage operation, using equation (23):

$$\text{At 500 Hz:} \quad \frac{\sqrt{P}}{E} = \frac{1.9 \times 1.75\sqrt{10.4}}{12\sqrt{(10.4 + 22/12)^2 + 6^2}} = .065$$

$$\text{At 5 kHz:} \quad \frac{\sqrt{P}}{E} = \frac{1.9 \times 1.75\sqrt{10.4}}{12\sqrt{(10.4 + 22/12)^2 + 60^2}} = .016$$

Response is down 12 dB at 5 kHz. This can be partially compensated by making the compliance of the front air chamber resonate with the moving mass as previously described. These calculations illustrate the fact that careful design is required to enable a horn speaker to span even as little as 3 octaves. The magnetic field is not readily increased greatly over the value used above, in the present state of the art. The radiation resistance at the diaphragm is determined by the diaphragm size and throat size. Increasing the diaphragm size raises the mass, and is the wrong direction to go. Decreasing the throat size helps but lowers the efficiency and can increase distortion, as will be explained later. Reducing the voice coil resistance is not an admissible expedient because it changes the impedance rating of the speaker, which is specified. This leaves reduction of moving mass, which is the means used in the design of tweeters.

Front Air Chamber

The front air chamber can be used to advantage to extend high-frequency response. However, with a simple coupling chamber between the diaphragm and the horn throat, interference can occur at the throat between sounds originating at different points on the diaphragm. Figure 11A shows two paths from diaphragm to throat. If the diaphragm is 2 inches in diameter and the horn throat is spaced 0.1 inch from the diaphragm, the distance from diaphragm to throat AB is 0.1 inch while the distance A B is 1.0 inch. Sounds arriving along these two paths differ in phase by 180 degrees at 7500 Hz, i.e., they oppose each other.

At other frequencies there are cancellations and reinforcements to various degrees. The frequency response is ragged and drops at high frequencies. Interference is minimized in high-frequency speakers by means of *phasing plugs* which provide equal path lengths from different parts of the diaphragm to the throat. One type of phasing plug is shown in cross-section in Fig. 11B. For good response to 20 kHz, where one-half wavelength is 0.67 inch, phasing plugs must be carefully designed and manufactured with great accuracy. This section of a quality tweeter somewhat resembles a fine watch in the impression it gives of precision.

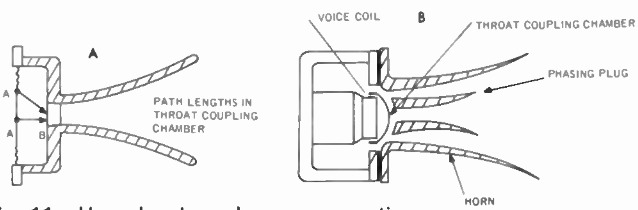

Fig. 11—Horn loudspeaker cross-section.

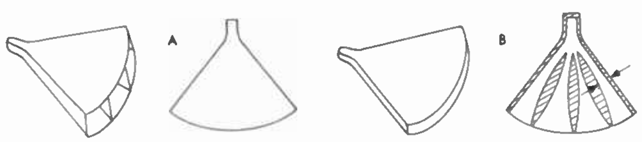

Fig. 12—A, sectoral horn; B, diffraction horn.

High Frequency Distortion

The maximum amplitude of vibration of the diaphragm is inversely proportional to frequency. The formula for the acoustic output of a horn loudspeaker whose diaphragm works into an acoustical resistance is

$$P_A = \frac{7930\, f^2 S_D^2\, d^2}{S_T} \tag{24}$$

where d = maximum excursion, one side.

For the theater horn loudspeaker described above to radiate 1 acoustic watt at 500 Hz

$$d = \frac{1}{f\, S_D} \sqrt{\frac{S_T}{7930}}$$

$$= \frac{1}{500 \times 28.3 \times 10^{-4}} \sqrt{\frac{3.14 \times 10^{-4}}{7930}} = .028 \text{ cm.}$$

This is a very small excursion indeed for a rather high power level, but it must be remembered that the clearance between the diaphragm and the phasing plug is usually very small also, so the excursion must be taken into account. At the lower frequencies the excursion may actually vary the volume of the front air chamber from nearly zero to almost double its normal volume. If high frequencies are present at the same time as the low frequency, the variable acoustic capacitance of the air chamber causes the low frequency to modulate the high frequencies. The amount of distortion depends on the power per unit area at the horn throat. At points along the horn away from the throat the sound intensity decreases and considerably less distortion occurs.

Distortion in high-frequency horn speakers due to nonlinearity of the suspension and of the magnetic system is less than in direct radiators because the diaphragm excursion is lower.

When distortion is referred to, one usually assumes that nonlinear distortion is meant, as evidenced by the generation of harmonics and intermodulation products. There is, however, an additional form of distortion in horn loudspeakers: phase distortion. The phase velocity with which a condition of maximum pressure is propagated through a horn is given by

$$C' = \frac{C}{\sqrt{1 - \frac{m^2 C^2}{16 \pi^2 f^2}}} \tag{25}$$

Since C varies with frequency, sounds of different frequencies travel through the horn at different speeds. The fundamental and harmonics of a given nonsinusoidal wave are thus displaced in phase; when $f = mc/4\pi$ the phase velocity becomes infinite; all parts of the medium in the horn move in phase as a unit, and the horn no longer functions as a horn.

Folded Horns

Bass horns must have large mouths and a slow rate of taper; consequently they are quite long as well. To reduce the space they occupy they are folded over themselves several times. Wide-range horns for public address installations also have their bulk reduced by folding. Typical PA speakers are folded concentrically to provide relatively long paths of expansion in compact form. These are called re-entrant horns.

There is one difficulty; sound does not go around corners readily except at low frequencies. As the frequency increases, when the dimension across the bend becomes an appreciable fraction of a wavelength, interference occurs in much the same manner as in a horn front air chamber without a phasing plug. The frequency response of a folded horn droops and is ragged in its upper frequency range. The effect can be reduced by minimizing the dimension across the bend and by shaping the outside curve so that it acts as a reflector. In high fidelity reproduction, folded horns are used only for low frequencies.

Directional Patterns

At low frequencies, for which the wavelength is comparable to the mouth diameter, a horn is almost omnidirectional. Its polar pattern is practically the same as that of a direct radiator speaker of the same diameter as the horn mouth. Above this range the horn becomes more directional but less so than a direct radiator of equivalent size, assuming the latter to operate as a perfectly rigid piston. The rate of narrowing of the polar characteristic with increase in frequency is relatively low.

In the frequency range referred to, the size of the horn mouth does not greatly affect the directional pattern, but the flare rate does. Slow rates of flare produce somewhat more directionality. Since the polar pattern depends on both mouth size and flare rate, different directional characteristics can be obtained in two planes at right angles to each other through the horn axis.

Horns used for high fidelity sound reproduction are usually not circular in cross-section because they are designed to provide dissimilar horizontal and vertical distribution. A widely used type of horn is the *sectoral horn.* The two sides are flat planes as seen in Fig. 12, and the horizontal faces are curved in the vertical plane, to provide exponential expansion. Horizontal distribution approximates the angle between the sides; the vertical pattern is determined by the flare rate and mouth size. If the vertical dimension of the mouth is small in comparison with the shortest wavelength radiated, the horn mouth tends to act like a small horizontal line source (a line of small speakers) and produces wide dispersion in the vertical plane. This is the reverse of what one would expect intuitively. This type of horn is called a *diffraction horn.* For good horizontal dispersion it must be mounted with its major axis vertical. Diffraction horns have been built with no vertical expansion, all the expansion being in the horizontal plane.

Rectangular horns are sometimes equipped with flat vertical radial vanes, making them resemble multicellular horns, of which more later. The vanes are mainly useful in stiffening the horn structure. If the vanes are tapered so as to provide in effect several adjacent horizontally adjacent exponential horns, they do affect the polar pattern. In one design, this type of vane is placed in the section of the horn near the throat and does not extend near the horn mouth. In effect there is a small cluster of horns within the main horn, acting to shape the wave front for the higher frequencies.

A cluster of horns with small mouths, designed to have a combined mouth that is essentially part of the surface of a sphere is called a multicellular horn. Ideally, such a horn produces identical sound pressure and phase in the spherical surface formed by the mouths, acting like a section of a uniformly radiating sphere. The distribution in the horizontal and vertical planes is controlled by the number of cells in each direction.

Multicellular horns are capable of producing extremely uniform directional patterns over a wide frequency range. Just below the frequency range in which the mouth dimension is between one and two wavelengths, directionality increases. Consequently, multicellular horns cannot be used for frequencies as low as can simple horns when mouth size is a determining factor, if uniform directional patterns are required. This means that, for a given low-frequency limit, this type of horn is larger than other types.

Simple horns can be given good polar characteristics by placing acoustic lenses in their mouths. One type of acoustic lens consists of a series of closely-spaced perforated screens so arranged that the outer parts of the wave from the horn are forced to travel a longer path than those closer to the center of the lens. The progressive delay in a directional outward from the horn axis bends the outer sections of the wavefront back, producing divergence of the wave. A series of plane or curved baffles can also be used to produce path-length differences. Diffraction around obstacles is another means of altering the shape of the wavefront.

The Loudspeaker/Living Room System

Roy F. Allison*

"What do people actually hear when they put on a record and sit in their favorite chairs, and why do they hear what they do?"

As part of a research project on loudspeaker measurement techniques, Acoustic Research recently measured the "frequency response" of the sound fields produced by loudspeakers in normal listening rooms. We wanted objective field data on real-life listening situations: what do people who buy high-fidelity loudspeakers, and put them where they will fit best in their living rooms, actually hear when they put on a record and sit in their favorite chairs? And (just as important for our purpose) *why* do they hear what they do? Which aspects of a loudspeaker system's performance are significant in determining the perceived frequency response, and which (if any) are not?

These questions arise, of course, because of the very significant differences in results obtained when loudspeaker systems are tested in different ways. The "frequency response" depends on the environment into which the speakers of the system radiate, the angle from the system at which the measurement is made, the distance of the microphone from the system, and even the time (relative to the input signal) of the measurement. It is not surprising that there is misunderstanding and controversy whenever loudspeaker measurements are discussed. Some of these differences may be clarified by the illustrations that follow. They show the results of tests on one particular model of speaker system under various conditions, with comments on each type of test. (To answer the obvious question in advance, it is an AR-3a system).

Tests made in an anechoic environment—either outdoors or in a chamber with completely sound-absorbing treatment on the walls—provide information on the direct radiation from the system but only at one angle from it at a time. Figure 1 shows the anechoic response of the individual speakers in the system, taken through the crossover network, at three angles: 0° (directly in front), 30° off the axis and 60° off the axis. The low-frequency part of the woofer curve was taken outdoors, since anechoic chambers are not perfectly sound-absorbent at very low frequencies. The mid-range and tweeter curves were taken in an anechoic chamber but with the speakers on large flat baffles to eliminate diffraction effects.

Figure 1 is only a starting point. This kind of response is never heard as direct radiation from a speaker system, because at and near the crossover frequencies there are two speakers, physically separated in the cabinet, radiating simultaneously. Their phase relationship for rays of direct radiation changes with the angle of the ray, reinforcing or cancelling in the region of overlap. This interference effect is shown in Fig. 2. These are anechoic chamber curves of all three speakers of the system, remounted in the cabinet and operating together. The cabinet's molding has been removed and the speaker mounting plate extended by a flat baffle. Response is shown at the same three angles as for Fig. 1. It should be realized, however, that while the curves in Figure 1 are typical of those that would be obtained for rays at the same angles in all planes, this is not true for the system curves in Fig. 2. The interference effects would be different for similar angles in different planes around the cabinet.

The first sound that reaches a listener's ears, regardless of the listening environment, is represented accurately by a response curve taken under the conditions that apply for Fig. 2. The exact curve that would apply depends on the angle of the listener with respect to the cabinet, of course. But this relatively simple situation does not last very long.

After a period of somewhat less than one millisecond, diffraction effects—reflections from the grille cloth molding and the cabinet edges—cause further perturbations in the response at any particular angle. This can still be considered "direct radiation" because, even though it is the result of reflections, it is caused by the cabinet and it is independent of the listening environment. Diffraction effects are visible in Figure

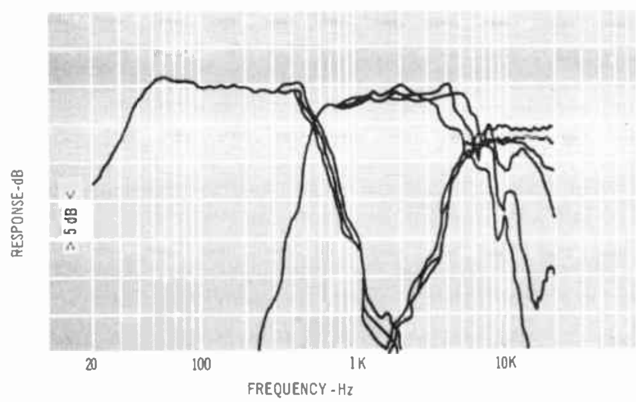

Fig. 1—Flat-baffle anechoic response of each of three speakers in the system, taken at angles of 0, 30, and 60 degrees.

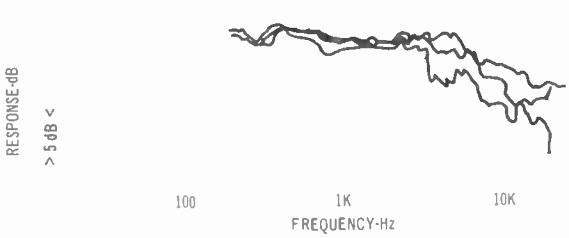

Fig. 2—Anechoic response of complete system in cabinet, but with grille cloth molding removed to minimize diffraction, at angles of 0, 30, and 60 degrees.

*Vice President, Acoustic Research, Inc., Cambridge, Mass.

3. These curves correspond to the ones in Fig. 2 except that the grille cloth molding has been reinstalled. Such curves represent accurately the sound field at listeners' ears during the time interval between the onset of diffraction (less than one millisecond) and the arrival of the first room reflections (3 milliseconds or so).

The room reflections build up in density (that is, the time intervals between individual reflections become shorter) and increase in total intensity, then fade away as the sound energy is absorbed by successive bounces from the walls and room furnishings. This reverberant field energy exists in significant amplitude for a period of ¼ to one second, depending on the reverberation time of the room and upon the original intensity. During this interval hundreds of reflections will occur, each of which affects the "response" of the instantaneous sound field at the listeners' ears. The sound pressure level of the reverberant field is quite uniform throughout the room. If the listener is more than four or five feet away from the speaker system, the reverberant field is significantly greater in amplitude than the direct field for most frequencies, regardless of the direction in which the speaker is "aimed."

The reverberant field is composed of sound energy that originates as radiation from the speaker system in *all* directions —not just the rays sent directly toward listeners. Therefore its "frequency response" is really the sum of the output at all angles (the acoustic power response of the speaker system), as modified by the frequency characteristics of the room itself.

How does the room modify the reverberant field response? Figure 4 shows the unmodified acoustic power response of this speaker system, with mid-range and tweeter level controls at maximum settings. This curve was obtained in a reverberant chamber—a small room deliberately made as reflective as possible, with minimum sound absorption. Its frequency characteristic is known and compensated in the measurement system, so that Fig. 4 is an accurate representation of the system's true power output vs. frequency. The room is not reliable below about 700 Hz, but the system is known to be omnidirectional below that frequency; thus its anechoic output at low frequencies can be considered to be representative of its acoustic power output. By comparing Fig. 4 with the results of the same kind of measurements made in actual rooms, therefore, the effects of the room can be seen.

We made such measurements at several locations in each of eight real-life rooms. They were the music listening rooms— the living rooms or recreation rooms—of eight AR-3a owners in the Greater Boston area. Neither the speaker systems nor the furniture was moved for these tests; the only thing we changed was the level control settings for the mid-range and tweeter units. They were turned to maximum for the tests, so that the results could be compared directly. The rooms varied substantially in size, shape, and "liveness."

Figure 5 is one set of curves for one of these rooms. The microphone for this test was placed eight feet from the left-channel speaker system and directly in front of it. Figure 5A is the curve obtained with the speaker cabinet in its normal position, facing the mike; 5B is the curve obtained by rotating the speaker cabinet 30°; 5C is the curve obtained with the speaker cabinet rotated 60°. Turning the cabinet, rather than moving the microphone, minimized the effect of room mode differences that would occur at different room locations. In this way we could change the frequency response of direct radiation reaching the microphone (as demonstrated in Figs. 2 and 3) and evaluate the effect on the total sound field in the room at the microphone location. The great similarity of the three curves of Figure 5 show clearly that the field at the location of the microphone is primarily reverberant—that the amplitude of direct radiation from the speaker system is far below the amplitude of the reverberant field. This was true for all normal listener locations in all the rooms.

Figure 6 is a curve obtained at another listening location in the same room, with both speaker systems operating and in normal physical orientation. This is a typical curve, about average in over-all shape and with a little more roughness than average. In general, we found that there were no sharp peaks or dips caused by room modes above 1 kHz. Whatever correction in general slope might be desirable could be done

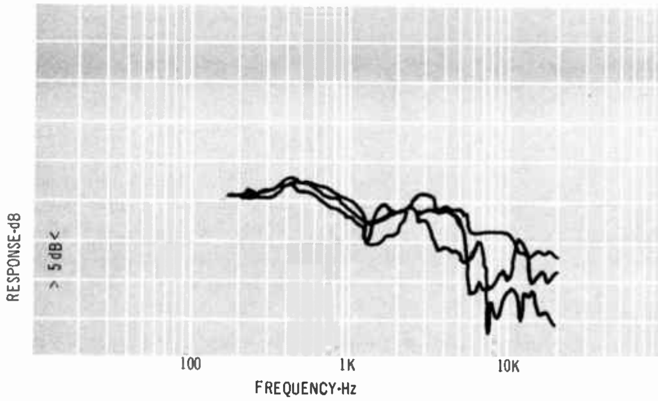

Fig. 3—Anechoic response of complete system in cabinet, with grille cloth molding, at angles of 0, 30, and 60 degrees. Diffraction would produce elevated output in 1.5-kHz region at some other angles.

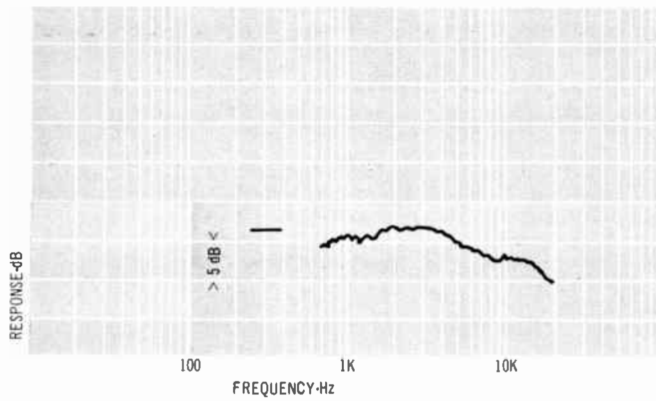

Fig. 4—Acoustic power response of the speaker system, measured in a reverberant chamber. Straight line at left shows relative woofer level.

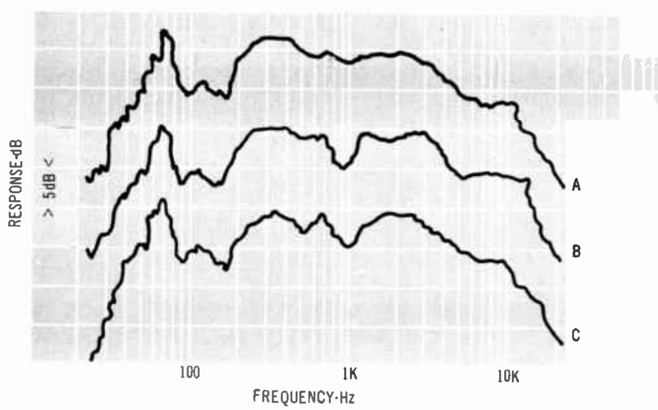

Fig. 5—**A**, Frequency response of loudspeaker and room at location eight feet from speaker system, with speaker aimed directly at microphone; **B**, same with speaker cabinet rotated 30 degrees, and **C**, same with speaker cabinet rotated 60 degrees.

quite accurately with a treble tone control or the level controls on the speakers. As for the room modes at low frequencies, notice the differences below 1 kHz between Figs. 5 and 6: correction for one room location would make response worse at the other location. It is difficult to see any justification for resonant narrow-band "room equalizers" if the speaker systems are good to start with.

One might argue that the relative amplitudes of the direct and reverberant fields are of no consequence. The direct wave reaches the listener first. Since directional perception is undeniably carried on by detection of very small time differences between the direct waves from two speaker systems, isn't it probable that listeners base their judgments of spectral balance also on the first-arrival sound wave, and ignore the reverberant field's spectral balance?

The first argument in response to that proposition is a negative one. Frequency response of the first-arrival wave is not affected by the room. *If* the direct wave's spectral balance were the perceived spectral balance, therefore, a speaker system would sound the same in any room; an orchestra would sound the same in any hall. Experience tells us that this is not so. As a positive test, however, we made binaural recordings (using a dummy head, with microphones built into the ears) of music played through speakers in several of the rooms. We rotated the speaker cabinet several times during each recording, as we did for the response curves in Fig. 5, thereby changing the direct sound's frequency response substantially.

Listening to these recordings with stereo headphones we were unable to hear any differences in spectral balance between the 0°, 30°, and 60° cabinet angles for any normal listener location of the dummy head. Slight differences could be heard if the dummy head was brought to within three feet of the speaker cabinet. Conclusion: listeners base judgments of spectral balance on the sum of the direct and reverberant sound fields, and for all normal listener locations the reverberant field predominates in amplitude. Therefore, the acoustic power frequency response of a speaker system is of primary importance. The direct radiation at any particular angle is important only insofar as it affects the ratio of direct to reverberant sound at a particular listener location in the room. By the same token, wide, uniform dispersion of output at all frequencies is necessary to achieve maximum uniformity in the reverberant field and assure its predominance at locations close to the speaker systems.

Another important question is this: what is the proper spectral balance of the reverberant field—what should be its frequency response? The first impulsive answer would be, "Flat, of course." If the goal is maximum accuracy in reproducing the concert-listening experience, that is the wrong answer, at least for recordings as they are now made and for live broadcasts using present microphone techniques.

The main microphones for recording sessions and live broadcast are always set up quite close to the instruments. Often they are very close indeed, particularly for soloists on the stage. As a result these microphones are in the "near field"—the direct sound predominates, and the microphones receive a spectrum of energy that is either flat or with accentuated high frequencies.

A concert hall audience, on the other hand, is well within the area of reverberant field predominance. That is true even for small intimate halls. The reverberant field of the average concert hall has a spectral balance that slopes down at the high-frequency end much more severely than that of the average living room. To duplicate at home the spectral balance of the sound perceived at a live concert, therefore, the energy put into the room by the playback system must also slope down at high frequencies. That slope should be tailored to make up the difference between high-frequency absorption in the hall and the home listening room.

Figure 7 contains two frequency response curves. One is a plot of the average spectral balance of four typical concert halls, measured (without audience) at orchestra-floor seats between 1/3 and 1/2 way back in the hall from the stage. The solid part of this curve is the actual empty-seat measurement; the dashed part shows the average result that would be expected with the audiences in place. The other curve is the average spectral balance we measured for 22 normal listener locations in eight living rooms with AR-3a speaker systems. It is clear that the best match would be obtained with both the mid-range and tweeter levels turned down well below maximum, and with a small amount of bass tone control boost or placement of the speakers in positions more favorable for bass output. These are average curves, however, and should be interpreted only as a place from which to start. In view of the actual variations found in both concert halls and home listening rooms, maximum realism for each record can be obtained only if one is willing to recognize that these slope variations do exist and to make liberal use of tone controls to correct for them. Æ

"It is difficult to see any justification for resonant narrow-band 'room equalizers' if the speaker systems are good to start with."

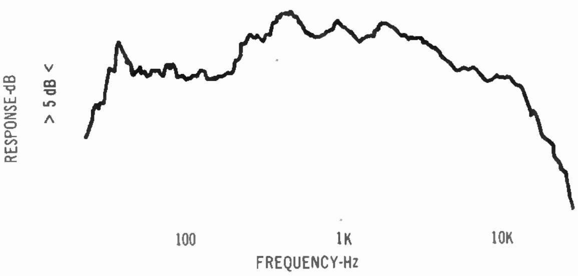

Fig. 6—Frequency response at another listening location, same room as Fig. 5, both speaker systems operating.

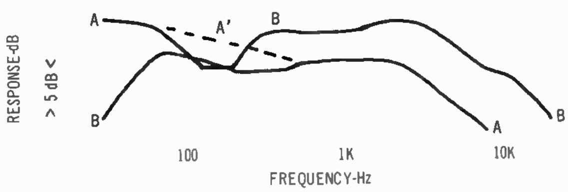

Fig. 7—A, Average spectral characteristic of concert halls, as actually measured without audience; **A'**, predicted result with audience, and **B**, average spectral characteristic produced by AR-3a systems at 22 listening locations in eight living rooms.

What Price Loudspeaker Response Curves?

Ralph West*

IT HAS OFTEN been said that loudspeaker response curves can be very misleading. For those who press for further explanation, one may say that even the closest examination of the curve tells very little about what the loudspeaker actually *sounds* like. Every manufacturer knows this only too well; he may well have learned to his cost, in the past, that it does not do to rely overly much on curves in deciding when to freeze the design and start production. A relatively flat curve, obtained after much sweat and tears, may turn out to be a shocking noise. Again, two very similar irregular curves (they usually are!) may produce entirely different sounds in practice, so much so that one may suspect the accuracy of the measurements.

Now, the frequency response of an amplifier, a recording system or a microphone, does give meaningful information as a rule, sometimes all that is needed. Add to that distortion and signal handling capacity, and assume reasonable noise and transient behaviour, and one has a pretty good picture of its sound.

But these are relatively simple devices in that the signal has generally traversed the device, from input to output, in a few microseconds, at most, a tiny fraction of one cycle of the highest audio frequency. As long as a device treats all frequencies alike, and amplifiers, etc. usually do, within a few percent, we are happy.

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Now a loudspeaker is far more complicated. It may take several milliseconds for the signal to reach all parts of its active area. Not only are these different parts very likely to vibrate independently, but they are all at different distances from our ears and from all the reflecting surfaces that surround them in normal use. The resulting sound then depends on just how these many independent parcels of sound add together or subtract. This varies with every change of frequency and with change of listening position, *i.e.* it is hopelessly complicated.

On the score of transients, the starting and stopping of signals, everything except the speaker is reasonably trouble-free. A good modern amplifier can start and stop in a microsecond or so, and a microphone in a few microseconds as its moving parts are small, light and easily damped. A loudspeaker, on the other hand, as it has to be so much larger and heavier—and stronger—to produce enough noise, has to be given so much energy to start it moving that there is a lot of stored energy to dissipate when the electrical input signal stops. Various patches of the cone are likely to go on wobbling long after the input has ceased. Moreover, none of this bad behaviour may show on the frequency response curve, which is a steady-state measurement.

The steady-state measurement is relatively easy to make and consequently is often performed, but transient behaviour is very difficult to measure—requiring a long-winded and tedious series of tests that could take days. The ear just listens and decides in seconds!

As engineers, we mistrust subjective assessments as they are liable to prejudice, peculiar likes and dislikes, and to variations from day to day. Objective measurements are far more reliable and repeatable, but still no use if they fail to give useful answers. In defense of the subjective assessment, one must realize that it involves also the services of an extremely complex computer, the brain, that can not only "measure" dozens of independent variables simutaneously, but also has a memory store several orders of magnitude larger than any man-made computer. It is therefore quite a formidable measuring tool.

A further look into some of the reasons why a response curve tells so little might be illustrated by the following. Place half a dozen people in a good row of seats in one of our large cathedrals to enjoy an organ recital and they would probably all agree that the sound was good, without adverse criticism. Now give them sound-level meters and graph paper, and get the organist to play slowly up the scale. The resulting graphs would be like cross-sections of the Rocky Mountains, and all different!

Of course the frequency response curve does show up *some* things we can hear. If the general level over large parts of the curve varies considerably—for instance, the average level for frequencies above 1 kHz is several decibels lower than that below 1 kHz—the sound will be dull and distant. If the level falls away steadily above 2 kHz (Fig. 1) it will sound muffled, whereas falling away below 1 kHz it will sound thin and shrill, crying out for considerable bass boost. These are matters of balance and they show up reasonably clearly.

Excessively high output over a narrow range of frequencies will also always produce an audible effect, possible because output is higher in this region, but more likely because there is also a resonance or series of reso-

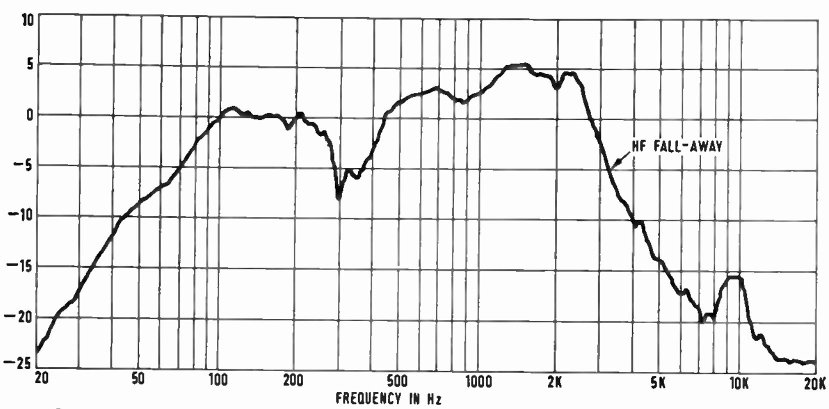


Fig. 1—Small sealed-box system with tweeter disconnected.

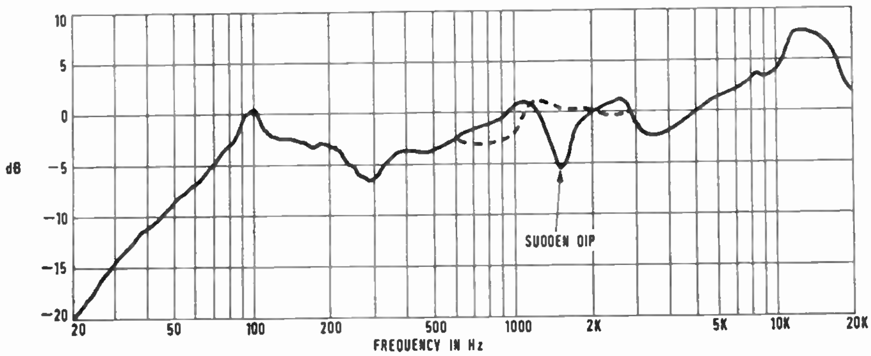


Fig. 2—Woofer and tweeter out of phase.

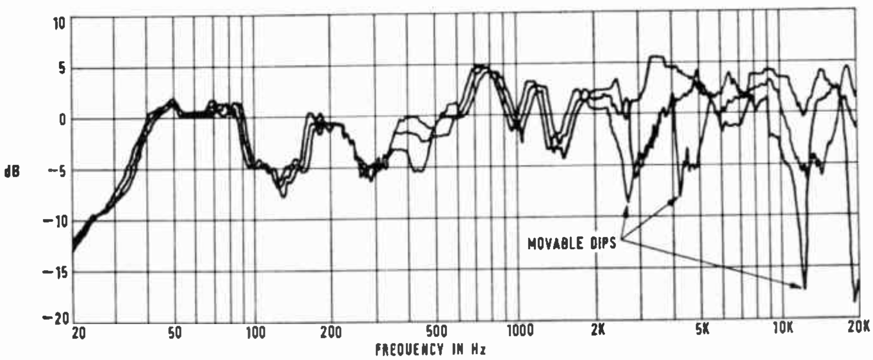


Fig. 3—Three microphone positions.

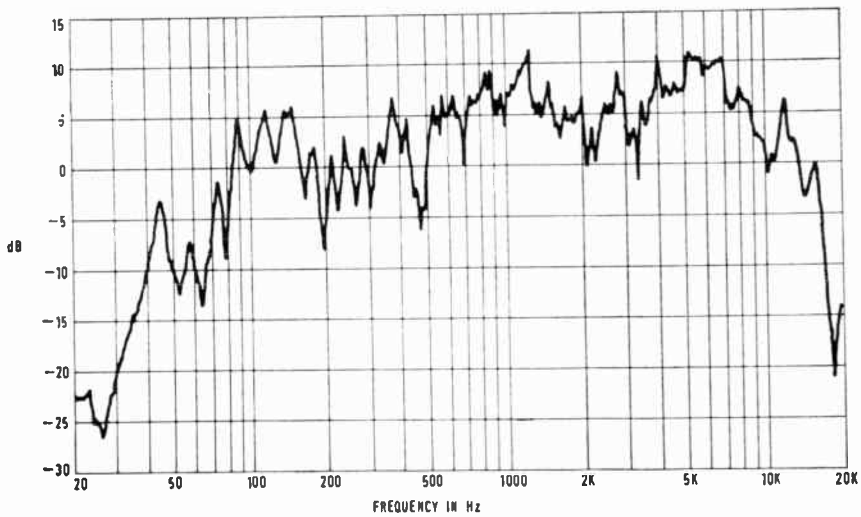


Fig. 4—Speaker "X" on left position.

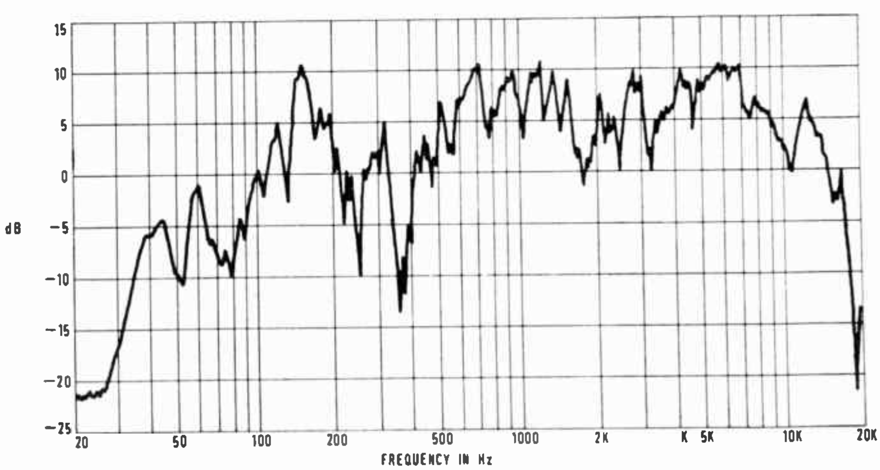


Fig. 5—Speaker "X" in right position.

nances close together. A resonance means a slow build-up of artificially exaggerated sound (not too serious) and a slow decay (very serious and one of the main dislikes of our ears, unless it is wideband natural decay like dying reverberation).

A large dip in a response curve is not quite so serious, as one cannot hear what is not there; but something will be heard, often a "hollowness," depending where it lies in the spectrum.

A large dip in the curve at the crossover frequency (Fig. 2) usually denotes a reversed phase between woofer and tweeter. On mono it may sound OK, but in stereo it often produces the most odd effects. Some instruments appear to pop over to the other side of the stage to play some notes, or an instrument may keep on altering its apparent width. Any sudden change in level on the curve is viewed with suspicion and often indicates something wrong that may produce an audible manifestation.

Any two-unit system can produce a whole series of different curves, depending on the exact position of the measuring microphone. Suppose the microphone is on the tweeter axis, it is then a little farther away from the woofer, and sound from the latter arrives a little later. If it is half a cycle later, it will subtract and the curve shows a dip. If perchance the two sounds are of equal intensity, and they should be at the crossover frequency, they will cancel, leaving nothing. Move the microphone slightly and this huge dip will disappear (Fig. 3).

Things like this, while they show up clearly on the frequency curve, cannot be heard in use. We listen with two ears, not one, the music is constantly changing, our head is continually slightly on the move, and in our domestic surroundings there are a whole host of reflections which complicate the pattern of variations still further. Even a single cone will still do this, as all parts of its area cannot be the same distance from the measuring microphone—and if they could be, it is unlikely that all these bits of the total area will be in step or in phase. (Only an electrostatic might manage.)

So, summing up thus far, the frequency curve of a loudspeaker has lots of wiggles that we shan't hear and a few wiggles we shall hear, but we cannot predict which. Only the big humps, hollows, and slopes are directly meaningful to a listener. Finally, looking at two different loudspeaker curves and trying to decide which is the better speaker of the two is almost a complete waste of time.

To the engineer, however, developing a new design, these measurements are

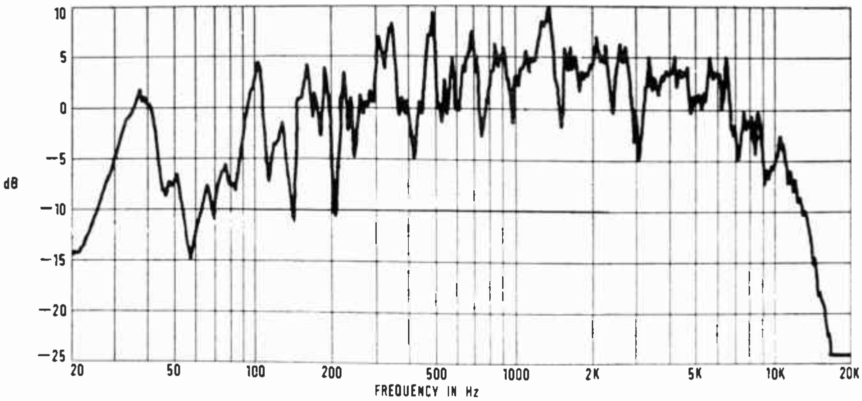


Fig. 6—Speaker "X" from Seat "A."

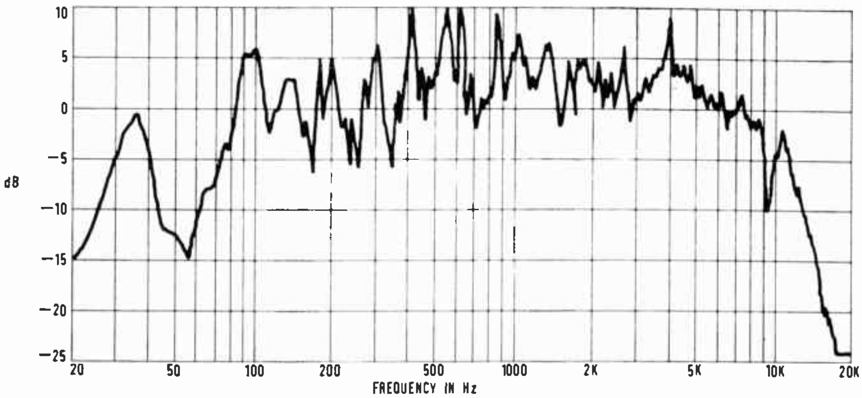


Fig. 7—Speaker "X" from Seat "B."

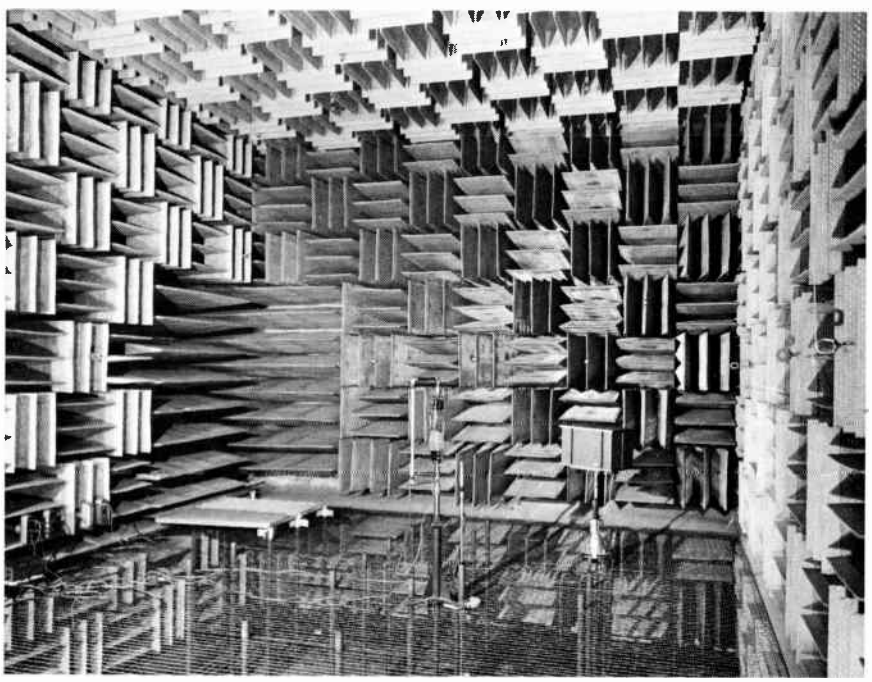


Fig. 8—An anechoic room for use in loudspeaker testing. Photo courtesy Jensen Sound Laboratories.

very useful. The curve will show, for instance, whether his design modification to increase or reduce the sound output at a particular point in the frequency spectrum has had the desired effect. If he is wise he will still listen to it to see if this modification has produced any undesirable effects that may not show on the curve. To the production engineer, regular measurements done under *exactly* the same conditions will show if standards are being maintained or if any changes are creeping into the product as supplies and staff, etc. change with time.

Having seen some slight justification for a loudspeaker response curve, how is it made? One merely puts a microphone with a flat response in front of the loudspeaker (one meter is the standard distance), varies the input frequency, keeping the voltage at the speaker terminals constant, and measures the microphone output. This sounds straightforward enough and does not take very long to do if one has a level recorder which plots the curve automatically. However, *where* is this done? The most obvious place is in typical listening room conditions. Speaker X, it had better be nameless (very non-standard), had the measuring microphone rigidly fixed to it with a light and acoustically transparent girder structure, to make sure we were always measuring the same speaker the same way. Placed in the left stereo speaker position, (Fig. 4) shows its measured behaviour. Placed on the right side, it gave Fig. 5. Out of sheer curiosity the speaker was placed in the center of the far end of the room and the microphone removed and placed, in turn, at ear level above two adjacent chairs from which listening normally takes place (Figs. 6 and 7). Any resemblance between Figs. 4, 5, 6, and 7 is (almost) purely accidental.

Quite clearly we are measuring the room more than the loudspeaker, especially in Figs. 6 and 7. This is why this measurement must always take place in non-reflecting surroundings. Ideally this is out of doors on a quiet, windless day, with the speaker and microphone about 20 ft. from the ground hung from a crane. This has been done, but is not too convenient, even if the weather permits, so at great cost we build anechoic chambers. These are very large rooms, preferably soundproof, with walls, floor and ceiling completely covered with a very thick layer of sound absorbing material. The most effective form of absorbent seems to be a mass of wedge-shaped members, with their thin ends pointing

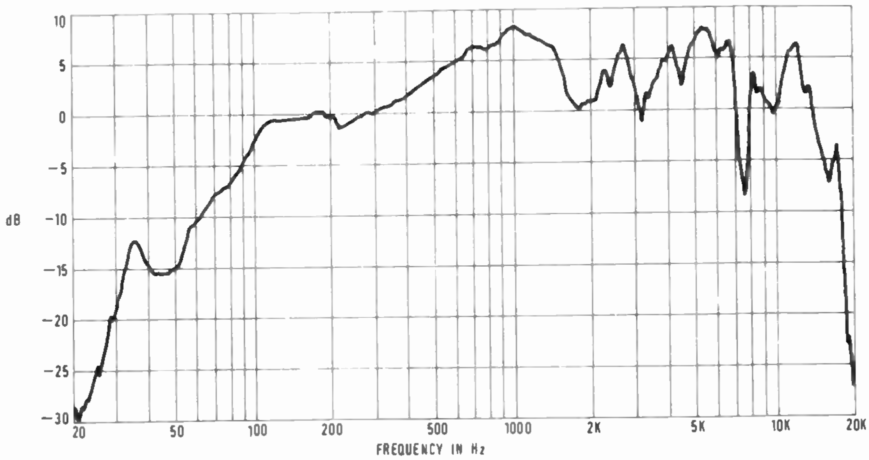


Fig. 9—Anechoic room.

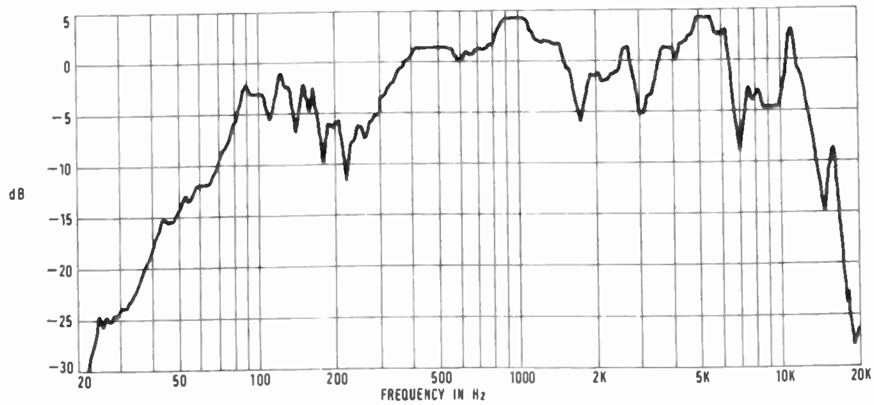


Fig. 10—Free-field (speaker on ground facing up).

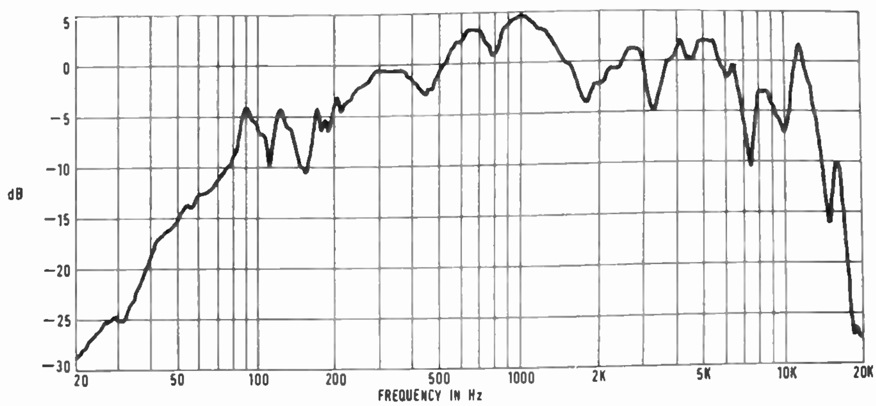


Fig. 11—Free-field (speaker raised 21 cm or 8¼ in.).

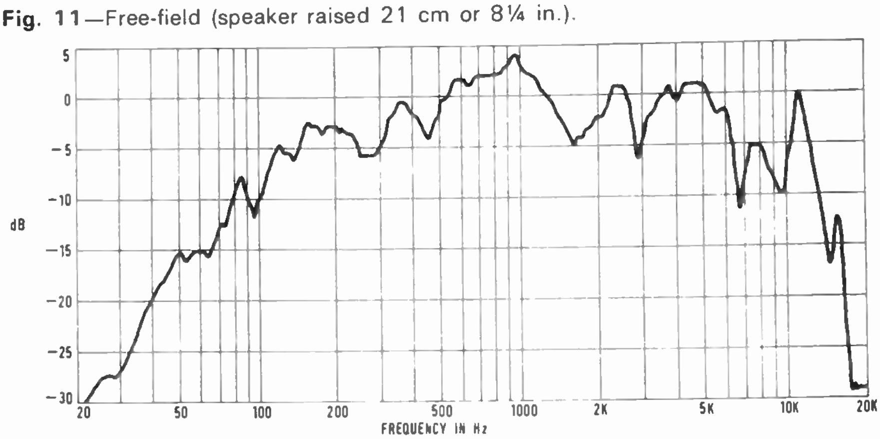


Fig. 12—Free-field (speaker raised 54 cm or 21¼ in.).

into the chamber. Special grades of polyurethane foam or rockwool in thin cotton bags are the most successful materials to date (Fig. 8). The length of the wedges determines the lowest frequency down to which practically 100% absorption takes place. To go down to 20 Hz the wedges would need to be about 12 ft. long! (I don't think there's even one in the world with 6 ft. wedges.) If the working space is large, then 5 ft. wedges give very small errors above, say, 30 Hz, and anyway one can always allow for a constant error.

Speaker X, with its attendant microphone, was taken to an anechoic chamber and gave the curve in Fig. 9. The little bump at 35 Hz is where the room is no longer absorbing most of the incident sound.

The poor man's anechoic chamber has to be the open air, and very useful results can be obtained quite simply. Ideally one should bury the speaker flush with the surface of the ground and hang the microphone directly above. Not wishing to dig up the lawn, or spoil the cabinet work, Speaker X was laid in the middle of the lawn well away from walls and buildings, and the curve of Fig. 10 resulted. This curve is better than Fig. 9 at the very low frequencies, but has a nasty series of dips in the 140-220 Hz range. This is due to some sound (from the woofer) reflecting from the ground (not too much as it's lawn) and arriving at the microphone half a cycle late and subtracting. Raising the speaker on to a low stool (21 cm) increases the path difference and cancellation takes place at a lower frequency: 100-200 Hz (Fig. 11). The three dips are thought to correspond to the three distances—over the side, over the bass end, over the tweeter end. Further raising on to a 54 cm stool and 185 cm step-ladder gave progressive lowering of the cancellation frequencies, and also less actual cancellation as the reflected sound is weaker due to greater distances travelled and more time for dispersion (Figs. 12 and 13). Fig. 13 is very close to the anechoic room curve: a few more feet higher and it would be acoustically superior. (*Memo:* must order scaffolding!)

Figure 14 shows a measurement taken with the speaker standing up normally on the same ground, the microphone now only 18 in. or so above the lawn. The bass end is now good, but the cancellation has moved up to 700 Hz and deepened. There is also evidence of reflections at higher frequencies too.

Apart from Fig. 14 the correlation above about 1 kHz is excellent for all

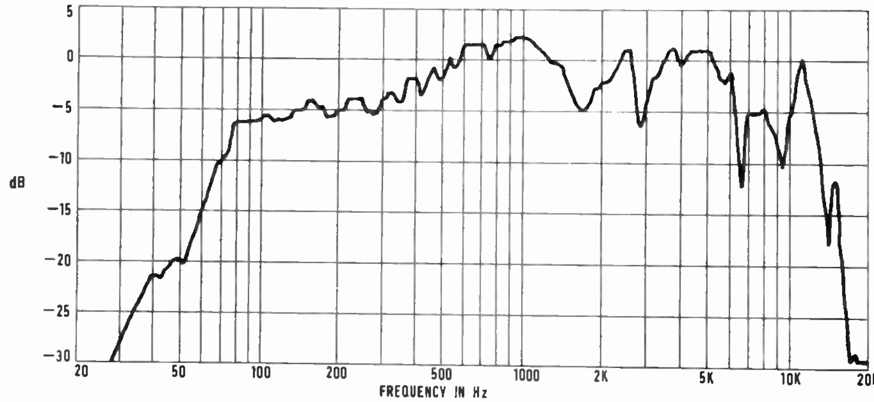


Fig. 13—Free-field (speaker raised 1.8 m or 5 ft., 10¾ in.).

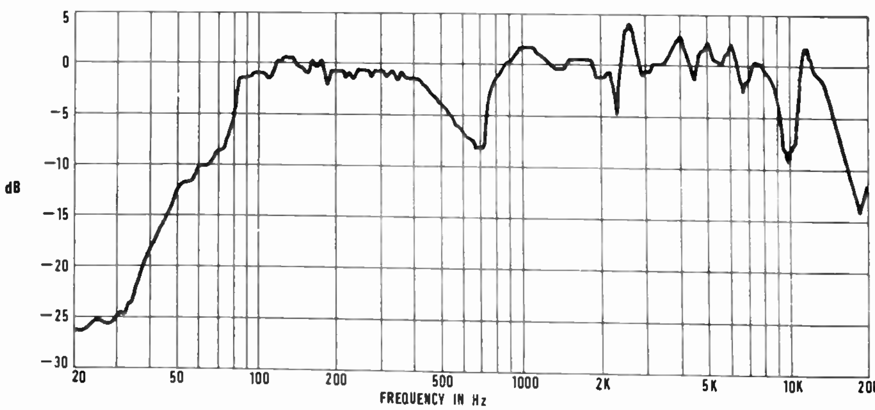


Fig. 14—Free-field (speaker on ground facing forward).

the other outdoor results. This is because the higher frequencies do not spill out over the cabinet edges like the low frequencies. With an omni-directional speaker, of course, one would have to hoist it aloft.

There is a small flaw in the validity of anechoic measurement, be it in a good chamber or hoisted aloft. This is bass loading. Aloft, at low frequencies the sound energy spreads in all directions. At home, our speaker is never far from floor and two walls, probably very close to one of them. Under these conditions, the bass is increased as the same power is now concentrated into something less than equal to a hemisphere. Comparing either Fig. 10 or Fig. 14 with Fig. 13, both show a good 3 dB higher level at 30 Hz. What should we do? And how would one measure something like a Klipsch corner horn? I give up! —No, there is more work to be done.

Editor's Note: AUDIO does not test speakers in an anechoic chamber with a mike three feet away. (Who listens that way?) Instead, speakers are measured in a typical listening room with "pink noise" (which avoids room effects), on and off axis, plus an average curve. Taken together with the other measurements, we feel it gives a reasonable review of the performance. Æ

Care of Records

Percy Wilson

NOTWITHSTANDING all the gadgetry that has been produced commercially, there are still many complaints that the records the public buys soon develop serious faults—pops, crackles, and warping, and may right from the beginning have eccentric holes. This even applies to records bought from specialist firms who guarantee that the records they sell will be unplayed and delivered in sealed envelopes.

There is little one can do about eccentric records—"swingers" we used to call them. Fortunately they are much rarer than they used to be.

Warping can be cured. The method is a little troublesome, but it is certainly worth while for anyone who really values his record collection. Two sheets of plate glass about 12½ inches square are needed. (Actually I use two circular sheets taken from the ports of old aircraft.) One of the surfaces of each of these should be scrupulously cleaned and polished. The record is then placed between the two and the assembly placed on top of a warm stove. If the glass sheets are thick, it is unlikely that the record will get too hot, though a temperature above about 150°F. at the record surface should be avoided. On the top of the upper glass a heavy weight can be placed. I used to use three volumes of the Encyclopedia Britannica. In about 12 hours the record can be removed and tested, on a turntable of course, for warping. In some cases the treatment may have to be repeated for a longer time.

The "pop and crackle" problem is more difficult. I have spent the best part of 10 years in research and have written two papers to the Audio Engineering Society describing my conclusions. Unfortunately, these indicated a remedy which was most expensive, since it demanded first of all an efficient method of wet cleaning and then a means of suction of the deposits from the record surface. I constructed about half a dozen pieces of apparatus to secure the result. My present system could not be marketed for less than about 250 British Pounds ($600), which as Euclid said, is absurd.

So I concluded that one must search for a simpler and less expensive procedure for avoiding the contamination. I think I have found it, and I am arranging with Metrosound to market the appropriate (and quite inexpensive apparatus.

First of all, though, may I explain the problem? Years ago, when Cecil Watts devised the Dust-Bug, and I applauded it in my *Gramophone Handbook* (now out of print), and Julian Herbage repeated my approbation in his Sunday morning broadcast on *Music Magazine*, Cecil wrote to me saying,

"What have you done to me? I have 5,000 letters to answer and no means to doing it."

But he proceeded to make a commercial device out of the Dust-Bug. All praise to him. He himself described it in his contribution to Gilbert Brigg's *Audio Biographies*.

His device dealt effectively with all the loose deposits which bedevil the records by electrostatic attraction. These are mostly fluff, but there are also fine particles of grit. These must be removed, and the Dust-Bug technique was aimed at minimizing the electrostatic discharge and removing the particles that broke through.

Our research, however, showed that there was a more subtle problem. The rotating record creates a sort of vortex which sucks down the air above the record, and then disperses it over the surface and shoots it over the edge. You can easily test this for yourself, as I demonstrated at the A.E.S. Fall convention in New York in 1964, by getting a piece of plastic tubing and inserting a metal tube about six inches long at the end (as a sort of cooling device). Then, holding this metal tube about six inches above the spindle of the rotating turntable, you fill your mouth with tobacco smoke and slowly blow it through the plastic tubing.

You will find that the smoke descends onto the turntable in a sort of exponential curve, and then shoots rapidly over the edge. A most intriguing picture!

This vortex, of course, drags down any minute particles that are suspended in the air, whether from smog, smoke, or aerosol activity (anti-fly, cosmetics, etc.) in the room.

These particles often escape the general vortex stream and deposit themselves mostly in two rings on the record surface—the principal one near the outer edge and a minor one at a radius of about 3½ to 4 inches.

Now these deposits, from their very nature, are sticky, and being so tiny in structure get right into the groove. Surface brushing does not move them, and if they are allowed to remain, they gradually harden and retain particles of grit which accompanied them on their vortex descent. Hence pops and crackles.

The present surface brushing devices do not remove this type of contamination. Wet cleaning with suction can do so, and the success is spectacular.

Is there any alternative to the wet-cleaning/suction process which is so

expensive? I have recently come to the happy conclusion that there is, provided one is careful. But it is progressive, and this means that it may take time in difficult cases.

One other qualification. Pops and crackles due to scratches cannot be removed. The sign of a scratch is that a pop occurs regularly at every revolution. So make sure that your inner sleeves are clean and free from grit. I prefer to slit the sides open so that one need not slide the record in and so rub the surfaces.

In all cases, new records as well as old, there are some rules that must be obeyed if the records are to be kept in good order. Following is a summary:

1. Never touch the recorded surface. Two methods of handling can be recommended.

A. In the open, place a second finger on the spindle hole with the thumb on the edge.

B. In putting the record onto the turntable and taking it off, hold it between the folds of a visting card or an old Christmas card.

2. Keep your turntable clean by regular brushing with a velvet or plush pad.

3. Never blow on a record to remove fluff. If you do, you will be sure to spray it with drops of spittle, and these are difficult to remove except by sophisticated wet cleaning.

4. Do not make the mistake of thinking that you can clean a record by a thorough washing with either detergents or under the tap. You can, indeed, but drying them afterwards is a difficult and risky business unless suction is used. Because of the grooves, a liquid adheres strongly by surface tension. When the record has been thoroughly wetted, a suction of 16 to 20 inches of mercury is needed to obtain a completely dry surface.

5. Be most meticulous in cleaning the stylus after every playing. Mere stroking is not good enough. If you were to examine the stylus through a deck microscope (say 100X), as I do, after playing a single side, you would be shocked to see how much debris it has picked up, even though you thought the record was clean. This has become much more troublesome with modern pickups which have short styli so as to reduce the tip mass. The former cartridges which we used to use for mono records had longer shanks, and the debris spread up them. Now it is all disposed near the tip. If it is not removed, it may settle down to the tip, and when you play the next side you will be using the debris as your playing point and of course distortion will result. Correspondingly, if you find the distortion has appeared during the playing of a record and suspect that the stylus has collected an unusual amount of debris, stop playing and clean the stylus.

To do this, a liquid must be used which will not affect the cement which fastens the stylus to the cantilever, and it must not have a high proportion of water. In addition, it must evaporate quickly. Moreover, care must be taken not to let the liquid spread to the other end of the cantilever, otherwise it will cause the moving magnet to rust.

A safe liquid is alcohol. Ethyl alcohol is the most efficient but is difficult to obtain. So as a general rule, one must fall back on isopropyl alcohol (which is the constituent of most of the commercial stylus or tapehead cleaners), or on vodka, which is almost pure alcohol, I am told.

The liquid should be used sparingly, preferably on a camel-hair brush such as children use in their painting lessons. Stroke the stylus gently from back to front, and, I repeat, do not splash the other end of the cantilever.

6. Never leave the record lying about out of its sleeve or on a rotating turntable unnecessarily. In the latter case, of course, the vortex action previously described will continue to put sticky deposits on the disc.

7. For storage, the records in their sleeves should be disposed vertically on shelves or in cases with slight but firm side pressure. There is on the market a cunning spring-loaded device which secures this pressure. But I have myself used two perhaps more amateurish ways. The problem, of course, is that the side pressure should not make it difficult for the record to be withdrawn from the stack.

The first device is to divide the stack by wedge-shaped partitions and provide the side pressure by the insertion of subsidiary wedges of opposite taper. (See Fig. 1.)

Insertion of the subsidiary wedges provides the required pressure; withdrawal of them releases the pressure and enables one to get hold of any record by grasping the empty top corner of its sleeve.

The second device is to have a partition in two halves joined together by double hinges. Such hinges are available at hardware stores. (See Fig. 2.) It will be clear from the diagram that their use enables the width of the double partition to be increased or reduced, and therefore to increase or reduce the side pressure.

8. So as to avoid unneccessary handling, do not make any attempt to store the records in any prescribed order on the shelves. For identification, number the sleeves either with sticky-back numbered tags or with a ball-point pen. Then have a loose-leaf notebook classified according to composer, artist or whatever you will. Suitable notebooks are of course not hard to come by.

9. Fortunately, in modern record-playing conditions, with diamond styli and playing weights of not more that 1 to 3 grams, stylus and record wear have become negligible. But do be careful not to drop your stylus by rough handling and do keep your equipment clean. Æ

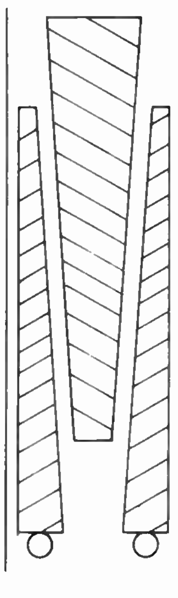


Fig. 1—Wedge method of obtaining pressure

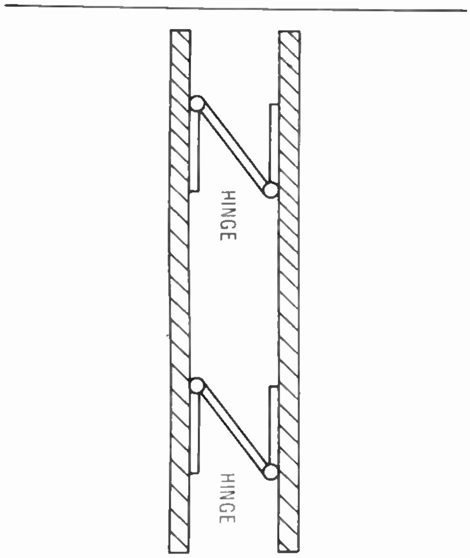
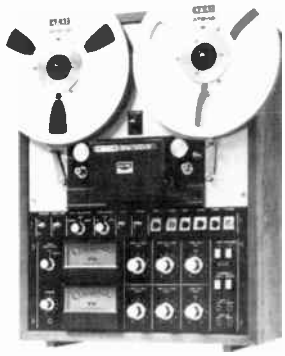


Fig. 2—Configuration of double hinges

Open-Reel Tape Recorders



Akai GX-400D

Crown SX-724

Dokorder 9100

Indicate speeds by letter code:

	A	B	C	D	E	F	G	H	J
15					x	x	x		
7½	x	x	x		x	x	x		
3¾	x	x	x		x		x	x	
1⅞	x		x	x			x	x	
15/16			x						

* at the highest speed of the machine

D after the price indicates the machine is Dolbyized.
All models solid state unless model number is preceded by (T)

MANUFACTURER	MODEL	Speeds (see letter code)	Power amp(s) built in ?	Max. reel size, in.	No. of heads	No. of tracks	No. of motors	Drive motor type	Drive to capstan	Frequency response * Hz to kHz. ± ? dB	Wow and flutter, % *	Signal-to-noise ratio, dB*	Fast wind, 1200 ft., sec.	Mic input Z, ohms	Rec'ing level indicator type	Dimensions, W x D x H, in.	Weight, lbs.	Price	SPECIAL FEATURES
AKAI	GX-400D	E	no	10½	4	4	3	Servo	Belt	20-26k -3	0.04	55	75	10k	2 Mtrs	18 x 22¼ x 8	75	1195 00	ADRS; rev. in rec & PB w/sens. foil; line/mic mix; SOS; Auto off/stop; PB vol. contl.; dual capstan.
	GX-370D	B	no	7	3	4	3	Servo	Direct	20-24k -3	0.07	55	60 75	10k	2 Mtrs.	17.7 x 9.8 x 19.9	57	799.95	Gl. & crys. ferr. hds. compute-o-matic; rev. rec. & PB; Auto off/stop; SOS; line/mic mix., direct func. chg. contl.
	GX-260D	B	no	7	4	4	3	Hys	Belt	30-22k -3	0.1	48	75	10k	2 Mtrs.	17½ x 18.6 x 8.9	52	579.95	Rev. rec. & PB; center capstan; rear pnl. vol. PB contl.; line/mic mix. SOS; auto stop.
	4440D	B	no	7	3	4	3	Hys	Belt	30-24k -3	0.12	50	90	10k	2 Mtrs	16.4 x 17.7 x 8.9	42	349.95	Dual mon.; SOS; SWS; sound mix.; auto off; pause contl.; tape sel. swit.
	4000DS	B	no	7	3	4	1	Ind	Belt	30-20k -3	0.15	48	120 150	5k	2 Mtrs.	15.9 x 7.6 x 12 4	25	269.95	Dual mon.; tape sel. swit.; SOS; SWS; Sound mix.; Auto off, pause.
	4400	B	yes	7	3	4	1	Ind	Belt	30-22k -3	0.15	50	190-180	5k 150k	2 Mtrs.	16.3 x 7 6 x 14	30.1	349.95	Conv. to deck w/frnt. pnl. swit.; SOS; SWS; sound mix.; dual mon.; auto off, pause.
	GX 280DDS 4 chan	B	no	7	4	4	3	Servo	Direct	20-25k -3	0.10	55	90	10k	4 Mtrs	17.4 x 10 x 20.2	49½	799.95	Gl & crys. ferr. hds.; dual mon., 4 ch. repeat cir., 2-ch. cont rev PB; auto off, stop; SOS; sound mix; pause w/lock
	1730DSS 4 chan.	B	no	7	4	4	1	Hys	Belt	30 22k -3	0 12	50	75	30k	4 Mtrs	16.8 x 9.6 x 14	19	399.95	Auto off; pause contl; 'phone jacks; 4-dig index ctr
	X-2000SD	A	yes	7	2	4	2	Ind	Belt	30 20k -3	0 2	48	75	0.5 mV	2 Mtrs.	10 6 x 13.8 x 18.3	49.4	549.95	Comb. reel-to-reel, cart. & cassette; crossfld hds.; auto shutoff/pause contl.; tk. sel.; tape lifter
	X-1810D	A	no	7	5	4	3	Hys	Belt	30-22k -3	0.08	50	60 75	100k	2 Mtrs	17.4 x 9.6 x 18	47	499.95	Comb. reel-to-reel & cart.; crossfld. hds., auto rev.; auto shutoff/stop.
	X-1800SD	A	yes	7	3	4	1	Ind	Belt	30-23k -3	0.12	50	150	0.5 mV	2 Mtrs.	13½ x 15 x 9½	37 4	399 95	Reel-to-reel & cart.; tk sel; auto shutoff/stop, tape cleaner; tape lifter in FF & rew.
ASTROCOM	407A	B	no	7	4	4	3	Hys	Idler	30-20k -3	0.06	60	45	10k	2 Mtrs	21 x 14½ x 10½	40	459.95	Auto rev
	711 4-chan.	B; F	no	10½	4	4	3	Hys	Idler	30 20k -3	0.03	65	60	Lo	4 Mtrs.			Under 2000.00	Headphone amps; mixing.
CROWN	SX-724	B	Opt.	10½	3	4	3	Hys	Belt	20-25k -2	0.09	60	45	350k	2 Mtrs.	19 x 9 x 15¾	45	1095.00	Dual mic, line mixing, 5 in. VU mtrs., also in ½ tk
	SX-824	B; F	Opt	10½	3	4	3	Hys	Belt	20 25k -2	0.09	60	45	350k	2 Mtrs	19 x 9 x 15¾	48	1495.00	Comp. logic, wal cab, ctr. opt, also in 2 tk
	SX-822	E	Opt	10½	3	2	3	Hys	Belt	30-30k -2	0 06	60	45	250 bal	2 Mtrs.	19 x 9 x 17½	53	1995.00	As above; also in ¼ tk. or 4-chan
	SX-744 4-chan.	B	Opt.	10½	3	4	3	Hys	Belt	20 25k -2	0.09	60	45	350k	4 Mtrs.	19 x 9 x 21	60	1995.00	8 mic inputs.
DOKORDER	9100	B	no	7	6	4	3	Hys.	Ind.	40-21k -3	0.06	55	95	10k	2 VU	17¾ x 15¼ x 20	55	799 95	Bi dir. rec; auto rpt PB; auto rev.; bias adj.; line/mic mix; tape/src mon; console.
	7500	B	no	7	6	4	3	Hys		25-23k -3	0.08	55	95	10k	2 VU	16⅞ x 6¾ x 17¾	41 9	599 95	Bi-dir. rec./play; auto rev/rpt PB; bias sel., elect contls.; echo; SOS; SWS; quick chg. head hsg.
	7200	B	no	7	4	4	3	4-p Sync		25 23k -3	0.08	53	95	10k	2 VU	16⅞ x 6¾ x 17¾	40.8	499.95	Cont. PB; auto rev.; bias sel.; tape/src mon.; line/mic mix; MBD heads
	7100	B	no	7	3*	4	3	Hys.		30-22k -3	0.08	55	95	10k	2 VU	16⅞ x 6¾ x 17¾	40	429.95	Bias contl.; src./tape mon.; echo, SOS, SWS; easy load design. *Interchg. 4-ch PB head
	7140 4-chan.	B	no	7	3	4	3	Sync	Ind.	30 22k -3	0 08	55	95	10k	4 VU	16⅞ x 6¾ x 17¾	40.7	569.95	Multi sync function; tape/src. mon.; bias sel.; echo, SOS; SWS; 4-ch. rec/play.

Open-Reel Tape Recorders

JVC RD-1555

Pioneer RT-1020L

Nagra SL Stereo

Indicate speeds by letter code:

	A	B	C	D	E	F	G	H	J
15					x	x	x		
7½	x	x	x		x	x	x		
3¾	x	x	x		x		x	x	
1⅞	x		x	x			x	x	
15/16			x						

at the highest speed of the machine

D after the price indicates the machine is Dolbyized.
All models solid state unless model number is preceded by (T)

MANUFACTURER	MODEL	Speeds (see letter code)	Power amp(s) built in ?	Max. reel size, in.	No. of heads	No. of tracks	No. of motors	Drive motor type	Drive to capstan	Frequency response ± ? dB Hz to kHz	Wow and flutter, %	Signal-to-noise ratio, dB*	Fast wind, 1200 ft., sec.	Mic input Z, ohms	Rec'ng level indicator type	Dimensions, W x D x H, in.	Weight, lbs.	Price	SPECIAL FEATURES
FERROGRAPH	704-AW	E; G	no	8¼	3	2; 4	3	ind	idler	30-17k ± 2	0.08	60	60	10k	2 Mtrs.	14⅞ x 16 x 8¾	37½	649.00	Model 704 ADW, Dolby $850.; 724 ADW $900.
JVC	RD-1555	B	no	7	4	4	3	Hys	Belt	40-18k ± 3	0.12	52	80	600	2 Mtrs.	16 x 8¼ x 17	34	499.95	Auto rev. w/direct cpld circuitry.
	RD-1553	B	no	7	3	4	3	Hys	Belt	20-24k ± 3	0.1	53	80	600	2 Mtrs.	16 x 8 x 18	33	499.95	2/4-chan· lo-noise & mon. swit ; retract pinch roller
	RD-1552	B	no	7	3	4	3	Hys	Belt	20-24k ± 3	0.1	52	80	600	2 Mtrs.	16 x 8 x 18	33	329.95	Solenoid batt.; SOS; flip-up head cover; retract. pinch roller.
	RD-1450	B	no	7	3	4	3	Ind	Belt	30-20k ± 3	0.1	52	160	600	2 Mtrs.	16 x 7 x 13	21	239.95	Tape/src swit ; auto stop; flip-up head cover; retract. pinch roller
	RD-1695	A	no	7	2	4	1	Ind	Belt	30-18k ± 3	0.13	52	190	600	2 Mtrs.	15¾ x 7¼ x 12¾	19	189.95	Rec. contls.; SOS; tape sel. swit.
	4RD-1401 4 chan.	B	no	7	3	4	1	Ind	Belt	30-20k ± 3	0.1	53	160	600	4 Mtrs.	16 x 8 x 18	27	449.95	2/4-chan ; lo-noise & mono swits ; retract pinch roller.
	4RD-1405 4-chan.	B	no	7	3	4	1	Ind	Belt	30-18k ± 3	0.1	52	160	600	4 Mtrs.	16 x 7⅝ x 13½	22	329.95	4 rec. contls.; tape bias.
MAGNAVOX	8877	A	no	7	3	4	1	Ind	Idler	50-15k ± 4	0.1	45	150	10k	2 Mtrs.	15½ x 13¼ x 7½		199.95	SOS; echo; noise suppr.; tape/src. mon.; 'phone mon.; tape sel.; bias freq. swt.
	8980	A	no	7	2	4	1	Ind	Idler	50-15k ± 4	0.15	45	180	10k	2 Mtrs.	15¼ x 12¼ x 7		159.95	'Phone mon.; 8981 same plus SOS; tape sel.; norm & hi O.D., $179.95.
NAGRA	SD	E	no	7	3	2		Servo		30-18k ± 2	*	70		200 50k	Mtr.	12.5 x 8.7 x 4.3	11½	2860.00	2 mic inputs; mixing; opt. remote; "D" cell pwrd.; *0.1% speed var.
	SNN	H	no		3					80-15k	0.1	−60		200	Mtr.	3.8 x 4 x 1.02	1.3	1364.00	Auto lev. contl.; mang. cell pwrd.
	4.2	E	no	7	4	1				30-20k ± 2	*	70		100k	Mtr.	12.5 x 8.7 x 4.3	11½	2310.00	"D" cell pwrd.; film sync.; AGC; *0.1% speed var.
PIONEER	RT-1020L 4-chan.	B	no	15	3	4	3	Hys syn	Idler	40-20k ± 3	0.08	55	90	20k	2 Mtrs.	17 x 9 x 16	46	649.95	4-ch. playbk.; snd. on snd.; rec. bias & equal. sel.; mic & lin input contls.; output level contls.; pause.
	T-6600	B	no	7	4	4	1	Hys syn	Idler	50-15k ± 2	0.12	55	110	50k	2 Mtrs.	17 x 7 x 17	28	349.95	Rec. & playbk. auto. rev.; pause; 4 dig. ctr.; l. & r. rec. levels; playbk. levels.
	T-6100	B	no	7	3	4	1	Hys syn	Idler	50-15k ± 2	0.12	55	110	50k	2 Mtrs.	15 x 7 x 15	26	279.95	Auto rev.; pause; l. & r. rec. levels; 4 dig. ctr.
RADIO SHACK	494 4-chan.	A	no	7	3	4	1	Ind.	Belt	50-18k	0.13	48	160	10k	4 Mtrs.	14¾ x 6⅝ x 16	27	299.95	
	999B	A	no	7	3	4	1	Ind.	Belt	40-20k	0.2	47		10k	2 Mtrs.	13¼ x 7⅜ x 16	20	199.95	
	909B	A	yes	7	2	4	1	Ind.	Belt	50-18k	0.25			10k	2 Mtrs.	24½ x 14 x 7½	26	239.95	W. 2 dyn. mics; detach. wing spkrs.
REVOX	A77 1104	B	opt	10½	3	4	3	Servo	Direct	30-20k +2 −3	0.08	62	60	hi/lo	2 VU Mtrs.	16⅜ x 7¼ x 14¼	34	899.00	w/amps & spkrs., $949.00; opt. rem. contl.
	A77 1102	B	opt	10½	3	2	3	Servo	Direct	30-20k +2 −3	0.08	66	60	hi/lo	2 VU Mtrs.	16⅜ x 7¼ x 14¼	34	899.00	W/amps & spkrs., $899.00; 7½/15 ips avail., $949.00; opt. rem. contl.
	A77 1134	B	no	10½	3	4	3	Servo	Direct	30-20k +2 −3	0.08	67	60	hi/lo	2 VU Mtrs.	16⅜ x 7¼ x 14¼	34	1099.00D	Opt. rem. contl.
	A77 1132	B	no	10½	3	2	3	Servo	Direct	30-20k +2 −3	0.08	70	60	hi/lo	2 VU Mtrs.	16⅜ x 7¼ x 14¼	34	1099.00D	Opt. rem. contl.

Open-Reel Tape Recorders

Sansui SD7000

Sony 854-4S

Technics by Panasonic RS-740US

Indicate speeds by letter code:

	A	B	C	D	E	F	G	H	J
15					x	x	x		
7½	x	x	x		x	x	x		
3¾	x	x	x		x		x	x	
1⅞	x		x	x			x	x	
15/16			x						

* at the highest speed of the machine

D after the price indicates the machine is Dolbyized
All models solid state unless model number is preceded by (T)

MANUFACTURER	MODEL	Speeds (see letter code)	Power amp(s) built in ?	Max. reel size, in.	No. of heads	No. of tracks	No. of motors	Drive motor type	Drive to capstan	Frequency response Hz to kHz, ± dB	Wow and flutter, %	Signal-to-noise ratio, dB*	Fast wind, 1200 ft., sec.	Mic input Z, ohms	Rec'ng level indicator type	Dimensions, W x D x H, in.	Weight, lbs.	Price	SPECIAL FEATURES
SANSUI	SD 700	B	no	7	4	4	3	Ind		20-20k ±2	0.06	60	65	50k	2 Mtrs	17⅛ x 10¼ x 21⅛	59½	549.95	Tone or foil strip auto rev
	QD-5500 4 chan	B	no	7	3	1	3	Ind	Hys sync	20-20k ±3	0.07	60	90	50k	4 Mtrs	16½ x 10¼ x 21¾	24.2	749.95	Rec/PB foil auto rev for rec PB
SONY	TC-270	A	yes	7	2	4	1	Ind	Idler	30-18k	0.12	50	150	Low	2 VU Mtrs	20¼ x 15¼ x 10¼	34.3	299.95	Rec EQ, scrape flutter filt., bass & treb tone contl., lid integ spkrs
	TC 353	A	yes	7	3	4	1	Ind	Idler	30-25k	0.12	55	150	Low	2 VU Mtrs	20½ x 10 x 13⅓	36.4	399.95	Echo, mag phono input, auto off, rec EQ, built in reel locks
	TC 630	A	yes	7	3	4	1	4 pole ind	Idler	30-22k	0.09	50	150	Low	2 VU Mtrs	17⅞ x 11⅝ x 20	46.3	449.95	Auto shutoff, mic & aux inputs, 2 stereo phone jacks
	TC-250	A	no	7	2	4	1	Ind	Idler	20-24k ±3	0.1	55	150	Low	2 VU Mtrs	15¾ x 14⅛ x 7⅜	18.1	199.95	Auto off, F & F hd., SWS, uni phase rec, built in reel locks
	TC 353D	A	no	7	3	4	1	Ind	Idler	30-25k ±3	0.1	55	150	Low	2 VU Mtrs	15⅜ x 13⅜ x 7⅞	16.9	249.95	Mic/line mix, built in reel locks, retractomatic pinch roller, pause w/lock
	TC 377	A	no	7	3	4	1	Ind	Idler	30-25k ±3	0.09	55	120	Low	2 VU Mtrs	16½ x 15½ x 8⅛	22.1	329.95	F & F hd, uni phase rec, built in reel locks, pause w/lock
	TC 458	B	no	7	4	4	1	Ind	Belt	20-30k	0.06	56	150	Low	2 VU Mtrs	15¾ x 8 x 16¼	32	479.95	Ultra hi freq bias, auto rev (cont.), TMS, built in reel locks
	TC 580	A	no	7	4	4	3	Servo	Belt	30-25k ±3	0.06	56	60	Low	2 VU Mtrs	17½ x 18⅛	37.6	579.95	ESP auto rev (cont.), elect speed chg, feather touch contls
	TC 640B	B	no	7	3	4	3	Hys	Belt	20-25k	0.07	55	90	Low	2 VU Mtrs	14½ x 9½ x 15½	35½	449.95	Echo, scrape flutter filt, TMS, rec EQ, reel locks
	TC 755	B	no	10½	3	4	3	Servo	Belt dual caps	20-30k	0.05	56	150 2400	Low	2 VU Mtrs	17⅛ x 8½ x 17¾	50.1	699.95	Log contl transp., F & F hd, mech mem cap
	TC 800B	C	yes	5	2	2	1	Servo	Belt	30-15k	0.1	48	100	Low	1 VU Mtr	12¼ x 10¼ x 4¼	11.1	279.95	Built in cond mic., 3 dig tape cntr, 3 pos mic sel swit
	TC 850 2	E	no	10½	4	2	3	Servo	Belt dual caps	30-25k ±2	0.03	57	130 2400	Low	2 VU Mtrs	17½ x 10 x 19¾	57½	995.00	w/4 tk play, built in SOS, mod constr, reel locks; TC 850 4 same w/4 tk & 2 tk play
	TC 277 4 4 chan	A	no	7	2	4	1	Ind	Idler	50-18k ±3	0.12	55	150	Low	4 VU Mtrs	15¾ x 15½ x 7¾	23.2	399.95	Quadradial rec play, 2 ch rec/play, reel lock
	TC 388 4 4 chan	B	no	7	3	4	1	Ind	Idler	20-25k ±3	0.09	55	75	Low	4 VU Mtrs	16½ x 8⅞ x 19¾	34.2	499.95	Spec pan pot when used as 2 chan deck, built in reel locks
	TC 854 4S 4 chan	E	no	10½	4	4	3	Servo	Belt	30-25k ±2	0.03	59	130 2400	Low	4 VU Calib Mtrs	17⅓ x 10 x 22	61.7	1795.00	Sync trak, closed loop dual capstan tape drive, Quadradial rec play
STELLAVOX (HERVIC)	SP7	G	yes	5-12 w/acc	4	1-2 4	1	DC Servo	Direct	30-16k ±2	0.04	70M 65S		200	Dual Mtrs	10½ x 8½ x 3	7	2195.66 to 2336.72	Mono/stereo, sync, interchgbl plug in heads
	SM7	G	yes	5-12 w/acc	3	2	1	DC Servo	Direct	20-28k ±2	0.04	72		200	Dual Mtrs	10½ x 8½ x 3	7	3528.00	A B or phantom mic pwrg, plug in modules
	SQ7 4 chan	G	yes	5-12 w/acc	3	4	1	DC Servo	Direct	20-25k ±2	0.04	66		200	Quad Mtrs	10½ x 8½ x 3	12	5731.58	Compl pwr indepndnt of a c, 4 chan rec

Open-Reel Tape Recorders

Tandberg 9000X

Tascam Series 70

15					x	x	x	
7½	x	x	x		x	x	x	
3¾	x	x	x		x		x	x
1⅞	x		x	x			x	x
15/16			x					

* at the highest speed of the machine

D after the price indicates the machine is Dolbyized.
All models solid state unless model number is preceded by (T)

MANUFACTURER	MODEL	Speeds (see letter code)	Power amp(s) built in?	Max. reel size, in.	No. of heads	No. of tracks	No. of motors	Drive motor type	Drive to capstan	Frequency response Hz to kHz, ? dB	Wow and flutter, % *	Signal-to-noise ratio, dB*	Fast wind, 1200 ft., sec.	Mic input Z, ohms	Rec'ing level indicator type	Dimensions, W x D x H, in.	Weight, lbs.	Price	SPECIAL FEATURES
TANDBERG	9000X	A	no	7	4	4	3	H.:	Idler	40 22 ± 2	0 07	58	55	200	2 pk Mtr	16⅛ x 15¾ x 7	34 5	699 50	Fingertip log contld drive sys
	3300X	A	no	7	4	4	1	Indc	Idler	40 22 ± 2	0 07	57	10 5	200	2 pk Mtr	16⅛ x 15¾ x 7	20 2	429 90	SOS, echo
	3600XD	A	no	7	4	4	1	Indc	Idler	40 22 ± 2	0 07	57	10 5	200	2 pk Mtr	16⅛ x 15¾ x7	20 2	580 00D	Dolby B Type noise red sys w/Dolby 67dB
	9200XD	A	no	7	4	4	3	Hys	Idler	40 22k ± 2	0 07	58*	55	200	2 pk Mtrs	16⅛ x 15¾ x 7	34 5	850 00D	Same as above but with Dolby 68dB
TASCAM	Series 70	F	no	10½	3	4	3	Hys	Indir	40 18k ± 2	0 05	55			4 Mtrs.	17½ x 10¾ x 23½	55	1950 00	Uses ½" tape, incl compl overdub swit, avail in 8 tk ½" version
TEAC	1250	B	no	7	3	4	3	Hys	Belt	40 18k ± 3	0 08	55	90	600	2 Mtrs	17 x 8 x 15	37	499 50	2 pos bias sw adj trntbl hght comb erase rec hd 2 play heads 1230 dame but w o rev play $399 50
	4010GSL	B	no	7	4	4	3	Hys	Belt	40 18k ± 3	0 08	55	90	600	2 Mtrs	18 x 10 x 18	45	599 50	Auto rev /play, ferrite hds, EQ/lev sw, dual lev bias, swit
	4070G	B	no	7	4	4	3	Hys	Belt	30 20k ± 3	0 07	58	90	600	2 Mtrs	18 x 9 x 18	51	699 50	Bi dir rec /play, ferrite hds EQ/lev swit, dual lev bias swit
	3300 10	B	no	10½	3	4	3	Hys	Belt	30 20k ± 3	0 07	58	90	600	2 Mtrs	17 x 9 x 17	45	589 50	"Quik loc" reel hldr, dual lev bias sw 3300 12 same plus ½ tk rec play $589 50 3300-11 sim w/"F" speeds $589 50
	2340 4 chan	B	no	7	3	4	3	Hys	Belt	40 18k ± 3	0 08	55	90	600	2 Mtrs	17 x 9 x 19	46	899.50	4 ch rec/play, simul sync mon dual lev bias swit
	3340 4 chan	F	no	10½	3	4	3	Hys	Belt	30 20k ± 3	0 05	55	90	600	2 Mtrs	17 x 9 x 21	50	969 50	Same as above
	6010GSL	B	no	7	4	4	3	Hys	Belt	30 20k ± 3	0 07	58	90	600	2 Mtrs	18 x 8 x 21	47	949 50	Auto rev play, ferrite hds dual EQ lev swit dual bias lev sw, 7010GSL sim but 10½" reel size $1149 50
	7030GSL	F	no	10½	4	2 4	3	Hys	Belt	30 22k ± 3	0 05	60	90	600	2 Mtrs	18 x 9 x 21	62	1099 50	2/4 tk play hd, ferrite hds dual EQ lev sw dual bias lev swit
TECHNICS BY PANASONIC	RS-1030US 4 chan	F	no	10	4	2 4	3	Hys	Belt	25 26k ± 3	0 08	58	60	600 50k	2 Mtrs	16⅝ x 11 x 23¼	70	999.95	2 chan rec/PB & 4 chan PB HPF heads, solenoid contl opt rem, bias sel, mixing
	RS 715US	B	no	7	4	4	3	Hys	Belt	10 22k ± 3	0 1	50	70	600 20k	2 Mtrs	17 x 9 x 20¾	53	549 95	Auto rev/PB, HPF heads bias sel, solenoid contls w/opt rem SOS, overdub
	RS 714US	B	no	7	3	4	3	Hys	Belt	10 22k ± 3	0 1	50	70	600 20k	2 Mtrs	17 x 9 20¾	53	499 95	Same feat as RS 715US except auto rev
	RS-740US 4-chan.	B	no	7	3	4	1	4 pole ind	Idler	30 20k ± 3	0 1	55	175	5k 20k	4 Mtrs	17 x 8¼ x 15¾	29	449 95	4 chan rec PB HPF heads, bias sel separation blend contls
	RS 736US	E	no	7	3	4	1	Hys	Idler	30 25k ± 3	0 0?	53	150	600 20k	2 Mtrs	17 x 8½ x 18½	33	329 95	Pause contl, bias sel, HPF heads SOS echo
TELEX	2001 W	A	yes	8¼	3	4	2	Hys	Belt	45 18k ± 2	0 13	52	80		2 Mtrs	19⅛ x 14½ x 8	48	799 95	VU mtrs auto off pause contl solenoid oper mon facil
WOLLENSAK (3M)	6150	A	no	7	3	4	2	Hys	Pulley	35 20k ± 2	0 12	54	90	2 2	2 Mtrs	16½ x 13½ x 6½	18	229 00	Record bias sel tape src mon
	6154 4-chan	A	no	7	3	4	2	Hys	Pulley	35 20k ± 2	0 12	51	90	2 2	2 Mtrs	16½ x 13½ x 6½	18	299 95	4 ch preamp deck

Choosing A Tape Recorder

H. W. Hellyer

OUR GRANDFATHERS did not have this problem. Choosing a tape recorder at the turn of the century was a simple matter—it was Magnetophon or nothing.

Now we have hundreds from which to make our choice, all—if you believe the advertisements—offering something special. The problem is, very often, to distinguish between those that offer what you want and those which are simply embellishing what may be a basically good model, but for which you pay more, get more, but use less.

First—read the reviewers. Read 'em diligently. Contrary to many opinions, these gents (myself among them, though not in these pages or even in this country) do their best to describe good and bad points of the equipment sent to them, adding a personal nuance here and there, which their own coterie of followers will learn to love or hate, and should always back their opinions with the hard bedrock of measured fact.

Second—if there is a point on which you are not clear, then ask! Ask the reviewer, ask the Editor, ask Joe Giovanelli; or for specific advice on tape questions ask Herman Burstein, ask the Editor, or, if you are prepared to wait for the Transatlantic carrier pigeons, ask Donald Aldous or me—but ask. . . . We are here to serve. We do not want you to waste your money, tuck the tape recorder under the bed and go back, disgruntled, to off-the-air programs of someone else's choosing. If there is a question about a switch function, a system matching, interpretation of technical terms, or any other suchlike bother, then we should be able to help you. But if it is a question of whether A is better than B, you may not be so easily served.

With tape, it depends on what you mean by "better."

Less noise? Well, yes, the noise reduction systems do have an effect and it is greatest with necessarily slow-speed cassettes.

For reel-to-reel machines it may seem less important. But tape hiss, the everpresent bugbear, is also relevant at higher speeds and on wider tracks. So the "add-on" noise reduction device is worth consideration.

Some tape recorders will already have the noise reduction facility built-in. Watch for those which add distortion or rob your recording of its dynamic range. There are several methods: One aim of most is to reduce the higher frequencies when the overall signal is lowest. Using the same argument as the more sophisticated Dolby, they sense signal level and apply a frequency-conscious stop filter. Some of these can be viciously potent; my advice is record and replay a familiar piece of music, then listen very carefully for those artificial rises and falls in ambient noise level that give the game away.

Some machines do it both ways, that is, they apply extra pre-emphasis during recording or can be switched to high-energy tape activation, then utilize a noise reduction switch on replay. Although these refinements are more often encountered on cassette machines, where practically *anv* improvement has to be beneficial (Ouch! Ed.), makers of reel-to-reel tape recorders are adding all sorts of similar sophistications.

Choice of a reel-to-reel tape recorder, as distinct from a cassette or cartridge machine, argues that one wants the choice of speeds or some greater track-switching flexibility than is currently available because of cassette head size and mounting restrictions.

Accepting that argument, you should look for the tape recorder that does what you particularly want and no more, but one that does it as well as can be.

Tape recorder makers will not love me for saying so, but there's a heck of a lot of apparatus gathering dust because its promise outlived its performance. New models come along: they glitter more, have a lot of inviting facilities, merit a very close look. . . . But when you look, remember that a "track-transfer, multi-sound, echo-flip device" may sound impressive when you show your friends, but may have cost you the dollars you could have spent in getting the simpler, higher-speed, better-performing machine.

Make your choice logically. Say: (1) What do I want to do with this thing? (2) What am I likely to want to do later? and (3) How about after that?

Point 1 is the most important as well as the most immediate. If your main work is vocal—interviews, reports of meetings, out-and-about effects—you need as good a microphone as you can get. Likely you will have bought it or still be drooling over the catalogs. OK, don't waste money on the machine with mic. Not unless *that* machine, for other reasons, seems the best.

General purpose work is a vague term, but we all know what it means. Monday you just had to catch Station Zee-Dee-Kay giving forth on Mahler's love-life; Tuesday was when Junior visited, with his newest-born just gurgling to be historied; Wednesday—well, wasn't that the day the Stag Club just "took off" and haven't you just got to prove that Joe had said what he did before carting your precious tapes and slides to the PTA social on Thursday for your annual lecture on the Lakes of Saskatchewan? General purpose means what it says: your machine has to withstand a bit of hard handling and maybe a can of beer in its innards.

So choose it, expecting the worst. But remember, that Mahler has to come over soft and clear, and you'll do that better at a higher speed, if quality of performance matters more to you than a miserly saving on the cost of tape.

Which brings up another point. Tape length, and, related to it, tape thickness. Calculate your probable program length and add a bit for fluffing. There are plenty of good tape length/time/grade tables around—no need to bore you with another. But let me remind you that, for example, a 1 mil polyester tape on a 7-in. reel (1800 feet length) can give you 45 minutes of recording if you are using half-track stereo at 7½ ips, to get the best signal-to-noise ratio and dynamic range from your machine.

Use a quarter-track machine (I don't like the term 4-track in this context, with so much multi-track studio stuff about) and you'll double the playing time, but have to invert reels midway through that doubled performance. The use of 3¾ ips would have gotten you the same playing time, but for that you would have sacrificed a couple of thousand Hz frequency response at the top end (and don't kid yourself, the hiss is not all up there!) and had to put up with a fractionally worse wow and flutter performance.

The point I am trying to make is that the real perfectionist will go for the highest speed and the widest track and to hell with playing time per spool. Bert Whyte was enthusing about the Nagra—only just breaking out into stereo, forsooth!, when we are all in a quadraphonic fever—and my personal memories of this machine are a blissful month of full-track 15 ips recording while its owner, an explorer friend of mine, laid low by some bug, left it in my loving care. As Bert says, if only we could run two units in sync, we could make such things as outdoor four-channel stereo recordings.

Coming down out of the clouds a while, let me say that, as an engineer, I have the utmost difficulty in choosing tape recorders. I am always more concerned with the design factors and the probable life, the true specification (not just what is quoted, which can be regrettably ambiguous), and the *basic facilities* of the machine.

This is all the more important when reel-to-reel recorders are considered. Mainly because the range of facilities becomes so much wider—and the ultimate quality realizable so much higher. (*That* should offend a few marketeers! Ed.)

I assume you are reading this because the virtues of easy loading, portability, and a library you can carry in your auto map pocket are not so important as those of wider frequency range, better speed stability, improved signal-to-noise ratio (Dolby notwithstanding), and a tremendous eruption of facilities.

Choosing your reel-to-reel recorder is truly a matter of matching your needs to your pocket. In a challenged market, you can be sure that the goods are competitive. If you are a true tape enthusiast, *now* is the time to invest. Quadraphonics, as far as you are concerned, is an impossible or impossibly costly conversion from two-channel stereo. If you want surround sound, get in there now. Makers are jostling for your custom and the technology is not likely to be outdated in the lifetime of your machine—all the headwork is going into broadcasting and discs.

The design commitment in tape recording today is toward getting more flux on the tape, tailoring circuitry to get less noise without losing or distorting signal, and—most significantly—motor control.

Where only one of the "big boys" had a servo-controlled motor a year ago, several are seeing the advantage of this device in 1972 and there are more waiting in the wings. It calls for a hunk of circuit, but that is no great problem in these days of ICs. It gives spot-on speed and makes for less hum radiation, besides cutting down weight.

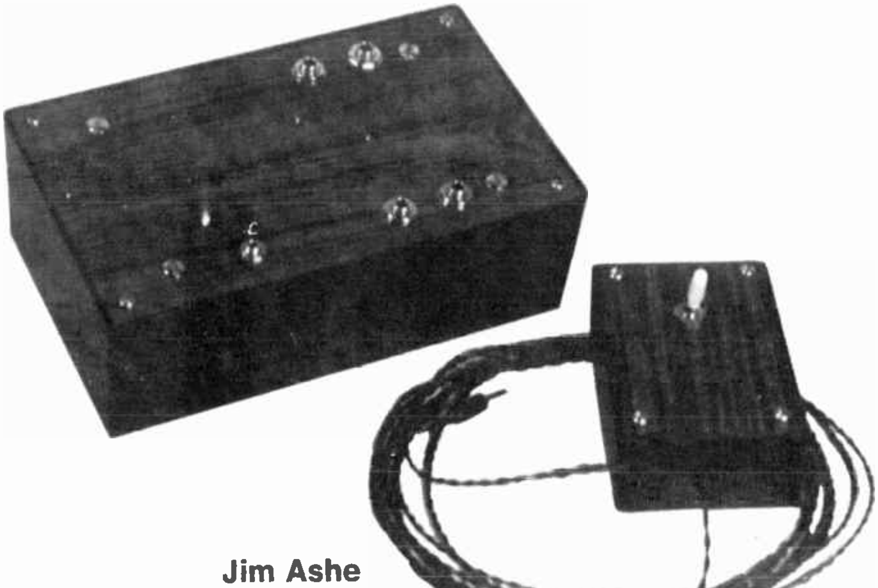
Automatic recording level control, once the specialty of the portable, is creeping into more and more of the larger models. So is single meter (peak-selective) indication. Both can have advantages in the right situation. Question is, is it yours?

How to choose from the welter of superlatives in the catalogs? The answer, as I said at the outset, is: Read your reviewers. Find whose opinion most matches your own experience or who consistently convinces you. Before making a choice, read as many independent reports on the model as you can. Shop around and try out that all-important factor, operability.

Yes, I *know* the word's not in Webster's. I coined it a coupla years ago when reviewing a portable which was the first I'd met where all the controls seemed to "come to hand." And *that* was a reel-to-reel machine, incidentally.

Make your choice the same way: Check the specs., read the reports, and handle it. If you are buying across the counter, be sure you get the chance to try before you buy. If you are buying mail order, well, you take a chance along with your discount.

Note that I've named no names? Deliberately. *You* have to do the choosing, friend: all I can hope is that my remarks may have sparked off a few ideas to help you. Only one thing is certain—the one you finally select will be the one you just can't quite afford. . . . Æ



Jim Ashe

THE KILLER THING

If you tape record radio and TV material, using your machine's PAUSE control for real-time editing, maybe you suffer from Taper's Lunge.

Taper's Lunge is marked by periods of gradually increasing nervous and muscular tension, released at moments keyed to commercials and station breaks. The release may be accented by rich Anglo-Saxon commentary. The Killer Thing described here is a simple device that reduces Taper's Lunge nearly to the vanishing point. This simple system improves off-the-air editing by reducing control motion to a hand-held switch operation.

Basically, it's a relay device in two boxes: one for control, the other for audio switching. From a modern engineering view, it's inelegant, obsolete, and works fine. See Fig. 1 for the schematic.

Because the input capacitors are grounded through resistors and the output circuits are connected to the same ground, connection into a circuit carrying accidental d.c. doesn't develop switching transients. The relay points develop no offset voltage, as would simple transistor or diode-switch circuits. Again, no thumps. The circuit through D1 dissipates inductive stored energy to avoid high-voltage pulses near the audio lines. With these considerations managed, the Killer Thing is as electrically unobtrusive as a piece of wood, except for doing its job. And it uses easily available parts.

A little thinking generates variations on the original. The Killer Thing could be assembled with only one relay, reducing costs. Or it might be wired as a cross-coupling switch that offers some interesting possibilities.

Construction

The finished Killer Thing is complete in two boxes with no power leads. One box fits the hand; it contains a switch and a pair of phone jacks. The other holds the relay and audio circuits and six pen cells for power. Standard audio leads patch the signal in and out of the relay box, and an inconspicuous cable completes a control connection to the hand switch.

A small 1 × 2 × 3 in. plastic utility box fits the hand nicely. It fits still better after its sharper corners are scraped and softened by a sharp knife or some sandpaper on a block. An SPDT switch goes in the box's aluminum panel and is connected to a working and a spare phone jack. Which is which depends upon user preference, but the spare is included because it might come in handy someday!

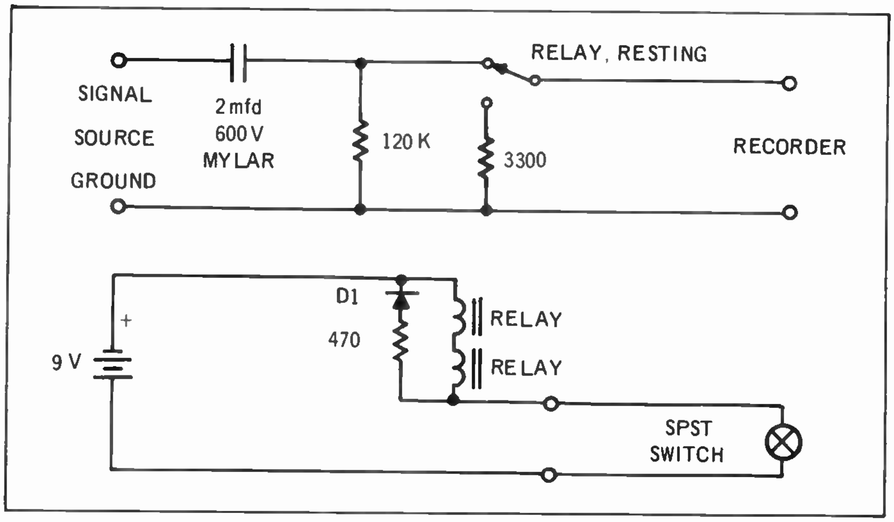


Fig. 1—The Killer Thing circuits are very simple. The audio switching circuit appears twice in the stereo version, once in the monaural version. Use whatever input and output connectors are convenient.

While the original Killer Thing uses subminiature phone jacks and plugs, if you're unskilled, choose the common miniature types. The subminiatures look good but assembly is pretty delicate. The original cable was 10 ft. long, made from one length each brown and black, #26 stranded wire. The two leads are adjusted and checked for equal length by observing equal sag when the cable is stretched across the room. Then a minute's spinning with one end chucked in an electric drill puts in a permanent twist. This makes a good, inconspicuous control cable. Since the cable carries about 10 milliamperes (one hundredth ampere) of current, wire size isn't critical. Lamp cord or lightweight speaker cable, for instance, would work very well.

The relay box begins as a normal 3¾ × 6⅛ × 2 in. utility box from Radio Shack. Four half-inch spacers hold a piece of perforated terminal board under the aluminum panel as shown in Fig. 2. This board is a piece designed as a box cover. It makes a good circuit board after edge filing and notching for wires and box clearance. A row of terminal clips toward one end of the board keeps the batteries from slipping into the electrical works, though nuts and bolts would work. Polyurethane scraps prevent rattling. The batteries are mounted in a plastic utility holder.

All parts go on terminal clips, with extra clips as relay terminals. This doesn't improve circuit operation but it's reliable, workmanlike, and looks good. Mounting parts on the clips before wiring helps avoid mistakes. All wiring is on the board side facing the aluminum panel; parts look at the box bottom.

The relays are Radio Shack #275-004, and it turns out these have odd threads. They take the same screws that hold covers on the Radio Shack utility boxes. A junked box from previous work donated two appropriate screws. For the kind of use the Killer Thing gets, the relays could have been held in place by their leads through the terminal board to clips.

Wiring goes fast, using #22 solid wire with thermoplastic insulation. Bare wire serves wherever there's no chance of a short. Wiring is uncritical, but good planning puts the relay-control wiring at right angles to any audio leads it crosses.

A wood-grained vinyl plastic covering, such as Contact paper, improves the raw aluminum panels. This requires care. First, all four panel edges are filed to make clearance between panel and box. After rough edges are scraped smooth, a piece of the pressure-sensitive vinyl plastic is cut an inch oversize (a half-inch per side) and gently removed from its backing paper.

The backing paper goes on the bench, panel over it, and the plastic is gently rolled onto panel and backing paper. Then the paper is pulled off, and if the work is bad, it's done over using new materials. When it's right, the four corners are cut at 45 degree angles, and the plastic edges folded tightly over the metal. A sharp knife trims out the holes. With a little practice this procedure makes very satisfactory, attractive panels.

Now the jacks go in the panel and get short leads before circuit board installation. Jack connections go through appropriate notches in the board to clips on the parts side, and the job is done. Ease on the panel screws, since the vinyl plastic wrinkles fairly easily.

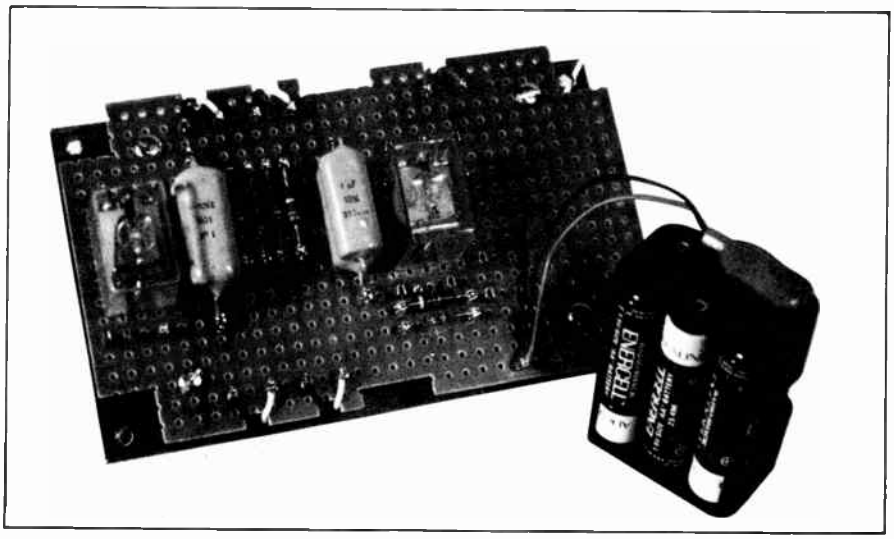


Fig. 2—A piece of perforated terminal board, designed as a box top, fits nicely inside if its edges are filed. Corners, box center rib clearances, and five notches for wires all can be cut with hacksaw and file or nibbling tool.

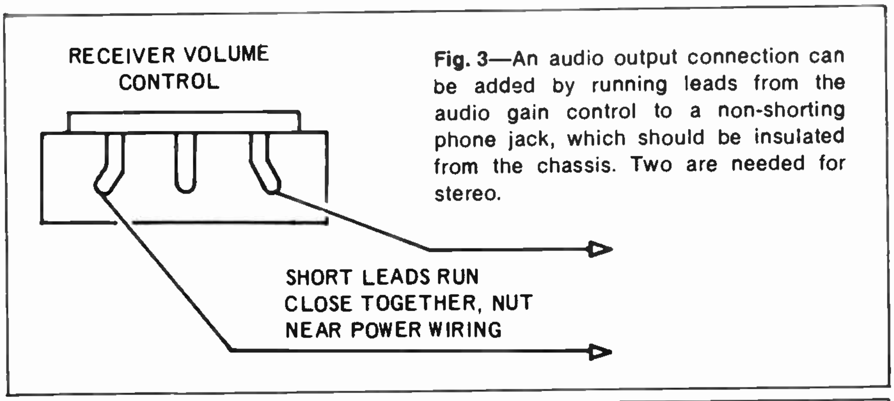


Fig. 3—An audio output connection can be added by running leads from the audio gain control to a non-shorting phone jack, which should be insulated from the chassis. Two are needed for stereo.

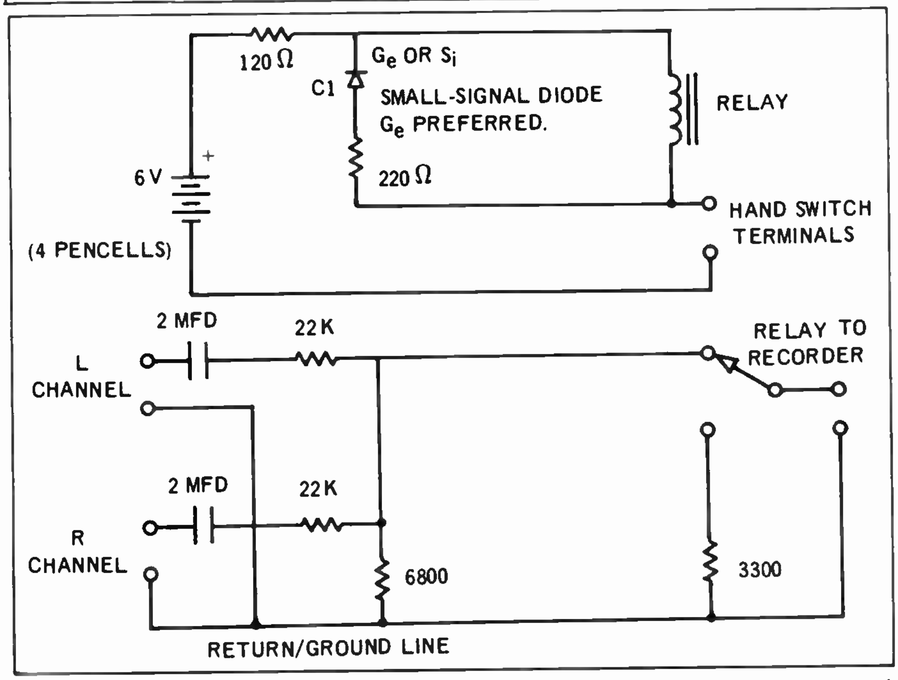


Fig. 4—A few parts changes are in order if one relay is omitted to make a monaural Killer Thing, and four pen cells provide adequate power. The mixer circuit for combining stereo channels into monaural could be placed in the Killer Thing box.

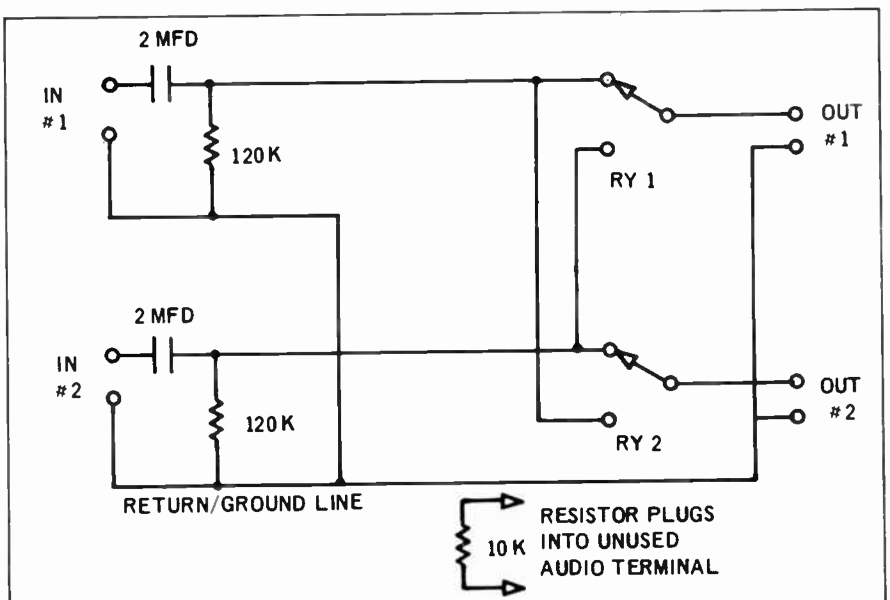


Fig. 5—The Killer Thing relays can be wired to cross-switch two audio circuits. An optional load resistor plugs into the unused jack to simulate a missing circuit.

Application

The Killer Thing goes between receiver and recorder without any special provisions. Where the receiver has no audio output connection, one is easily added by running leads to the audio gain control. (See Fig. 3.) A nonshorting phone jack is installed out of the way with no connection to anything except audio signal and return leads. It should be insulated from any metal chassis. If the audio is fairly strong at this point, as it usually will be, no special care is needed in wire placement. Leads should be short anyway. In use, which hand-switch positions are AUDIO-ON and AUDIO-OFF depend upon which hand-switch jack is used. Assembly with one jack and changing wire connections in the hand unit would work as well at a tiny parts-cost saving.

Battery life approximates battery shelf-life. An estimate based on Eveready's Battery Book comes out roughly 130 hours of actuated-relay time. Assuming a commercials-duty factor of 15 per cent or 9 minutes/hour, and two hours work per day, then the pen cells should go 430 days. But few users will follow that regular a schedule. If you make a Killer Thing right now, you can put in new batteries every even-numbered New Years Day.

The Killer Thing in use gives a fine sense of power. It edits accurately, since it works easily. When removing extended "clutter" (several commercials end-to-end, an increasing practice), the recorder can be PAUSED and started up again when the verbiage approximates an end. Then the audio is returned as the program material resumes.

Figure 4A shows how one relay can be eliminated for a monaural Killer Thing. Figure 4B shows how two stereo channels can be brought to one monaural, with equal emphasis to each. This circuit could be fitted in the relay box. It loses some signal but it's simple, and usually there is signal to spare. This arrangement might be used to store large amounts of monaural on a four-channel tape..

Another variation uses the Killer Thing for cross-switching. This is versatile since a modified Killer Thing can switch an input between two outputs, or two inputs into one output. That will come to you in a moment. Figure 5 shows the circuit, which unfortunately doesn't mix with the original Killer Thing design. Note the optional load resistor so the unconnected circuit thinks it's still loaded. This Killer Thing variation could be interesting to modern music workers who want to unobtrusively reverse channels. ●

THE CHANGING FACE OF OPEN-REEL

market and product trends

by G. L. Taylor

Once it was thought that the magnetic tape had a strong chance of replacing the vinvl record in the hearts and collections of music lovers. Today some probably still hold that hope—or fear—but their numbers are few. A glance into almost any current publication devoted to audio arts will reveal the continued prevalence of Record Power.

Yet, magnetic tape does hold an important place in the audio industry, due to its special capabilities which no record on the market today can equal.

The electronics industry is a fast-changing one. Innovation follows innovation: new products appear, old ones fade away. As products change, they attract new and different markets. Conversely, as the markets change, the products must change to accommodate them. This importance of market, then, is a primary concern in the study of trends in open reel tape recorders.

A strong demand for the versatility and high performance standards of open reel tape will always exist in the professional audio industry: broadcasting studios, recording studios, production companies, tape duplicators, wherever a high-quality master is necessary. These same characteristics continue to make open reel an invaluable part of computer operation and the sciences.

As an example of the latter, in a major motion picture, a bank of open reel tape decks is used in a program to translate the speech of dolphins. As with all good fiction, this movie has its feet planted firmly in fact: the ability of dolphins to communicate and their high level of intelligence long ago intrigued real-life scientists sufficiently enough for them to initiate similar programs at marine laboratories and universities around the world.

More importantly, the same qualities that keep open reel an entity in the professional market will serve it well in the consumer market, for a variety of reasons.

1. Increasing sophistication of music lovers. There are those who demand the best simply because it is the best. Others have weighted cassettes and cartridges in the balance and found them wanting. Many have started out with cassettes or cartridge units and are ready to step up. All of these people recognize and salute the well-deserved superior status of open reel.

2. Versatility of open reel. Cassettes and cartridges have as their principal advocates those whose primary interest is listening to music. For them, versatility is likely to be the ability to cut out commercials when recording off the air—if indeed they're interested in recording at all.

However, those whose interests extend to creating their own source material often demand more: three heads for editing and monitoring; special effects capabilities, such as echo, sound-on-sound, sound-with-sound, mic-line mixing. These are functions of open reel units.

3. Transition of music from vicarious listening experience to gut-level participatory experience. It used to be that composing and performing were arts that look incredible talent and many years to develop. Consequently, direct participation was limited to a relative few. No longer. Today music has become a more direct, personal expression of raw emotion open to nearly everyone with the urge to express. It is a trend characterized not so much by talent as by sincerity. This is evidenced by soaring sales of musical instruments in the past decade and by the tremendous outgrowth of new musical groups, each performing its own music. The fact that some have hit fantastic success almost overnight has encouraged countless others to leap on the music bandwagon.

An open reel tape recorder is vital for any group of burgeoning composer-musicians hoping to succeed. For creative editing and special effects are as much a part of this music as electric guitars, earnest vocalists and arcane lyrics.

4. Money to spend. All of the foregoing market trends would be worthless in an indigent society. People can pay more for open reel equipment simply because they have more to spend. Despite unemployment and inflation, disposable income seems to be on the rise.

These changing market trends have naturally had their effect upon products. But the situation is not without a certain irony, for open reel, in its increasing sophistication, seems to run counter to the prevailing trend toward simplicity in tape recorders today.

For dozens of years, a prime objective of the tape recorder industry was miniaturization, heralded in by the transistor and the integrated circuit. This trend was reflected in May 1967, in a series of prognostications published by AUDIO MAGAZINE on the occasion of its 20th anniversary. Tape recorders, said industry leaders, would get smaller and start using slower speeds, with better results than the higher speeds of older models.

Coincidentally (?) the same issue of AUDIO carried an article on the Phillips cassette and two rival forms of the tape cartridge: Fidelipac, a standard of the broadcast industry, and Lear-Jet, created by that manufacturer of personal jet aircraft in conjunction with Ford Motor Company. In 1967, none of these configurations were viable competitors for either the record or open reel tape. Their FI was admittedly far from HI: the article considered them adequate for automotive use, but not "sonically attractive enough to warrant the attention of the serious audio buff."

Today, of course, that's all changed. Serious audio buffs by the hundreds are buying cassette and cartridge units—and not just for their cars. Technological advances have apparently fulfilled the prophesies of the industry's augurs. Innovations such as Dolby and chromium dioxide tape coatings have put cassette performance, even at 1⅞ ips, on a par with records. Four-channel sound and a wide selection of software have made cartridge units a force to contend with in the home entertainment field. Ultimate miniaturization is nigh. Already palm-sized cassette units exist that give better performance than behemoth open reel models of yesteryear. At least one of these, SONY's TC-55, features a select switch that enables the unit to capture music with adequate, though not high, fidelity. Further refinements will undoubtedly follow.

So the cassette and cartridge are now respectable. Anyone with doubts can check prices, for one thing in this ever-changing industry remains constant: good performance still costs more than mediocre performance. The result is that cartridge and cassette tape units have taken over the market previously held by low- and medium-priced open reel units.

Consequently, as cassette and cartridge units become more compact and convenient, open reel units have been forced to become more complex and versatile. Features designed to provide professional-quality performance are finding their way more and more into consumer models.

1. Higher speeds. Paradoxically, as one faction of the industry moves toward the slower speeds predicted in 1967, another faction seems intent on defying destiny. Units with 15 ips speed settings are on the increase—corresponding to demand for the wide frequency response that's a function of these higher speeds.

2. Large reel capacity. To circumnavigate the reduced playing time that results from using higher speeds, an increasing number of recorders are being designed to accommodate 10½ inch reels. In addition, large reel capacity at the lower speeds can be useful to provide greatly extended continuous play.

3. More sophisticated drive systems. Uneven tape motion has long been a bugbear of the tape recorder. Now the sophisticated audiophile can reduce it and its attendant ills—wow-and-flutter and poor tape-to-head contact—with any of a growing number of 3-motor tape decks on the consumer market.

Innovative motor design is also helping to solve the problem of uneven tape motion. Typical is the servo-control motor, which prevents capstan speed variations due to normal voltage or load changes. (Figure 1.)

Connected directly to the capstan motor (A) is a frequency generator (B), whose frequency is dependent upon the motor RPMs. The frequency generated is relayed to a servo control board (C) which compensates for variations in the speed of the capstan motor by either increasing or decreasing voltage (D). The result is highly accurate motor speed and consequently, consistent tape motion.

Further sophistication can be found in yet another feature previously restricted by cost to the professional, but now making its way to the consumer: closed loop dual capstan tape drive.

In this system, two capstans isolate the tape from external vibration and abnormal reel movement by forming a "closed loop" of tape around the head assembly. Two current 3-motor tape decks featuring the system, SONY models TC-854-4S and 850, can attest to its efficacy with extremely low wow-and-flutter specifications of 0.03%.

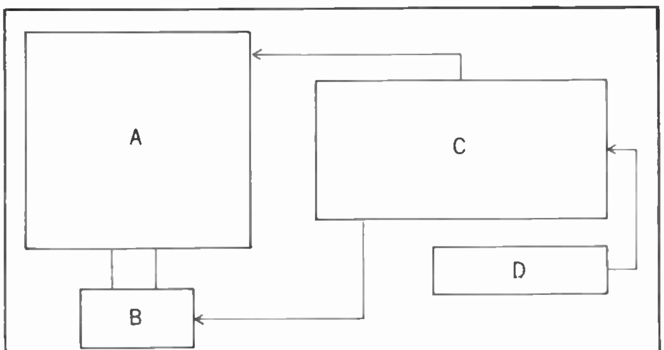


Fig. 1—Block diagram of a servo-control motor.

5. Ferrite heads. Much has been written concerning the hardness of the ferrite head and its resultant ability to resist abrasion. There are other advantages. For example, the higher internal resistance of the ferrite material used in their manufacture allows ferrite heads to be molded of one solid piece of material. By contrast, the permalloy head must be built up of laminations in order to cut down eddy current losses. The single surface of the ferrite head material can be lapped to a sharper, more precise edge than can the laminations of the permalloy, permitting more even pole pieces, and ultimately a narrower head gap. (Figure 2.) The effect on frequency response is obvious.

Natural stereo and 4-channel separation is virtually dependent upon even pole pieces and straight headgaps to prevent sound drop-outs from phase shifting. It's not surprising, then, to see the ferrite head making an appearance on the more sophisticated open reel units.

What else does the future hold? Knowing that the tendency is toward professionalism, we can make educated guesses—or think wishful thoughts. Chromium dioxide tape for reel to reel? Why not, at least at the slower speeds. Refinements in ferrite heads will certainly aid the cause. So will special lubricants to reduce friction, and its resultant abrasion.

Abrasion is one of tape's most insistant drawbacks. Hopefully, the time will come when it's eliminated altogether, or at least reduced to a nominal level. A more efficient method of reading the magnetic pattern on tape would help—especially one that does not depend on tape-to-head contact. As incentive, consider the recently developed phono cartridge that uses a photo cell instead of a needle.

This is a transitory period for open reel. It has lost sales to cassette and cartridge simply because its new identity and market are not yet firmly established. That time is coming.

To be sure, the market for the sophistication provided by open reel is smaller than that for the convenience of its plastic-enclosed cousins. But price differential minimizes that problem—and furthermore, each purchaser of a cassette or cartridge unit is a potential step-up to open reel.

One thing is certain: no matter how far cassettes and cartridges progress, open reel is still the ideal they're measured against. As long as that's true, open reel will remain an important factor in the consumer audio industry.

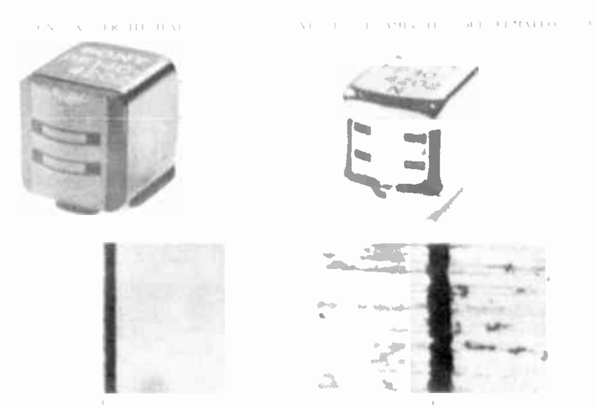


Fig. 2—Ferrite head vs Permalloy head, demonstrating the effect of laminations upon gap width and core smoothness.

The Tape Guide

All About Tape Recorder Equalization

Herman Burstein

FROM READERS' QUESTIONS and things that appear in the popular audio literature, it seems that tape recorder equalization is less well understood by audiophiles than its importance deserves. The mystery tends to be compounded by the variety of equalization characteristics necessitated by an assortment of tape formulations and tape speeds, as well as by the occasional promulgation or advocacy of new equalization standards.

Therefore the Tape Guide seeks to explain what tape recorder equalization is all about—why it is needed; how it is achieved; how it can be modified to optimize the interdependent requirements of extended treble response, low noise, and low distortion; the nature of the NAB equalization standards, and how equalization is affected by such things as tape speed, tape formulation, use of the Dolby system, etc.

For the most part the discussion assumes that tape speed is 7½ ips. Despite the greatly improved performance obtainable at lower speeds, 7½ ips is the NAB (National Association of Broadcasters) standard speed and is the one generally preferred for high quality home recording. (In fact, some home recordists prefer 15 ips.) In any event, what we have to say applies in principle to all tape speeds.

It is further assumed that the tape machine has: (1) clean and demagnetized heads so that no treble losses occur due to poor tape-to-head contact or magnetization; (2) heads in perfect azimuth alignment (gaps exactly at right angles to tape length) to eliminate treble losses owing to incorrect azimuth; (3) bias set for approximately minimum distortion at mid-frequencies, specifically at 400 Hz.

Why Equalization Is Necessary

Suppose that the tape machine we have just described has no equalization circuits to alter frequency response, and that it is employed to record and play a tape at 7½ ips. Further suppose that input signals of *constant level* are recorded throughout the 20-20,000 Hz range and then played back. A meter connected to the output would typically show the machine's record-playback response to be quite similar to Curve ABC in Fig. 1: Record-playback response climbs steadily at virtually 6 dB per octave (20 dB per decade, to be precise), reaches a peak around 3,500 Hz, and drops substantially thereafter.

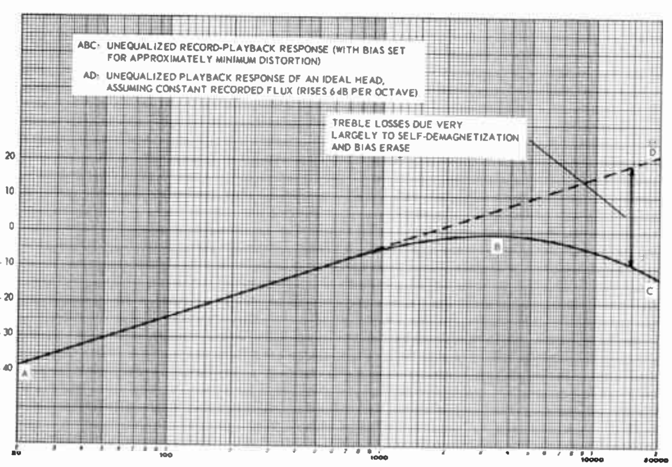


Fig. 1—Unequalized record-playback response of a tape recorder at 7½ ips.

Clearly, bass boost is needed to compensate for the drooping bass portion AB, and treble boost is needed to compensate for the drooping treble portion BC. That is the role of equalization—to provide bass and treble boost made necessary by the inherent nature of the tape recording process.

To help us see why Curve ABC is the way it is, Fig. 1 supplies line AD, which is the response of an "ideal" (perfect) playback head if a tape were recorded flat; that is, if the tape contained recorded flux of equal magnitude at all audio frequencies. At this point let us carefully note an important distinction between *applying* a flat signal to the tape and *recording* a flat signal on the tape. Losses, which we describe shortly, take place in the treble range of the recorded signal. However, line AD assumes there are no such losses so that a flat signal is recorded on the tape. In sum, AD is the playback response of an ideal head if the tape is *recorded* flat.

AD rises steadily at 6 dB per octave because the head is a velocity device. That is, the head is an electromagnetic generator with a voltage output proportional to the rate of change of the magnetic field of the tape. The field changes at a rate corresponding to the audio frequency. Hence the voltage output of the head is proportional to audio frequency. For example, at 10,000 Hz the playback head produces twice as much output as at 5,000 Hz, and 10 times as much as at 1,000 Hz. (That is why we say line AD rises 6 dB per octave or 20 dB per decade, since 6 dB represents (very nearly) a doubling of voltage, and 20 dB represents (exactly) a 10-fold increase).

Beyond approximately 800 Hz the record-playback curve fails to continue its 6 dB per octave climb due almost entirely to magnetic losses that occur *in recording* and become more severe as frequency increases. These losses are of two kinds, self-demagnetization and bias erase, and we shall return to them in a moment. There are also slight losses—especially slight at higher speeds—attributable to the playback head. Winding capacitance of the playback head may result in treble loss. And there may be some treble loss due to gap width (the wider the gap of the playback head and the slower the tape speed, the greater the loss.) But with a playback head that is well made and boasts a gap as narrow as 40 or 50 microinches, capacitance and gap losses at 7½ ips are usually quite negligible—on the order of 1 dB or less at 20,000 Hz. Thus we are principally concerned with recording losses described by the terms self-demagnetization and bias erase.

Self-demagnetization refers to the fact that the recorded signal on the tape in effect consists of a series of bar magnets end to end. The higher the frequency, the more bar magnets are recorded per inch of tape, so that each magnet is necessarily shorter. But the shorter the bar magnet, the closer together are its north and south poles, and the more their opposing magnetic fields tend to cancel; that is, the signal tends to self-demagnetize. In sum, with increasing frequency the strength of the recorded signal—the amount of magnetic flux on the tape—tends to weaken.

Bias erase is a side-effect of the high frequency signal, typically 75,000 Hz or higher, which is fed in moderate amount to the record head to minimize distortion and generally maximize the amplitude of the recorded signal; this is called bias current. In much greater quantity, about 10 times as much, the oscillator current powers the erase head. Unfortunately, bias current in the record head has the deleterious side effect of also accomplishing erasure—not nearly as effective as the erase head, but erasure nonetheless. Bias erase increases with frequency because the higher frequencies penetrate the tape less deeply and hence are more vulnerable to an erasing field. Altogether, bias current produces treble loss; the larger the bias current (for reduced distortion), the greater the treble loss.

The magnitude of the magnetic losses in recording is indicated by the interval between Line AD and Curve ABC. Recall that AD is the response of an ideal playback head in the absence of recording losses, and that an actual high quality playback head is close to ideal. Therefore the interval between AD and ABC represents recording losses. For example at 15,000 Hz the interval shows a loss of about 30 dB. Roughly 20 dB of this may be ascribed to self-demagnetization and the other 10 dB to bias erase.

Equalization for Flat Response

A major goal of high fidelity is of course flat frequency response, output signals having the same relative levels as

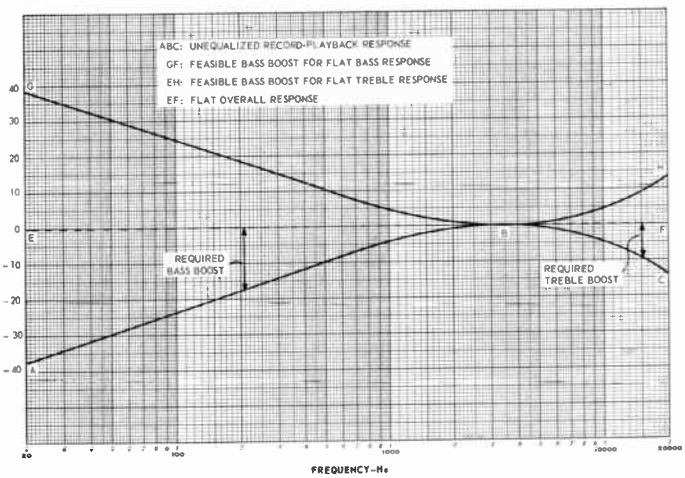


Fig. 2—A feasible pattern of equalization for a tape recorder at 7½ ips.

the input signals at all audio frequencies. In other words, for constant level input there should be constant level output in the range of approximately 20 to 20,000 Hz. Hence flat response is represented by a straight horizontal line, such as EF in Fig. 2.

Record-playback Curve ABC from Fig. 1 is repeated in Fig. 2, and a feasible scheme of equalization for flat response is straightaway evident. The interval between EB and AB may be interpreted as bass loss, and therefore represents the bass boost needed for flat bass response. Similarly the interval between BF and BC represents the needed treble boost. Accordingly, GF is a suitable bass equalization curve, rising in a fashion that mirrors the decline AB. And EH is a suitable treble equalization curve, rising in a fashion that mirrors the decline BC. Together, GF and EH complement record-playback Curve ABC to produce flat response.

Figure 2 is a workable scheme of equalization and something fairly like it is used. However, matters are not all that simple. In addition to flat response, high fidelity has low noise and low distortion as major goals. For reasons connected with improving the signal-to-noise ratio, actual equalization (the generally employed NAB standard) is a modified version of GF and EH in Fig. 2. But we must postpone, and pave the way for, discussion of NAB equalization in order to deal first with the question of where equalization takes place in a tape machine so as to best serve the triad of goals—flat response, low distortion, and low noise.

Where Equalization Takes Place

We begin with an important observation. Figure 1 shows great treble losses in recording, reaching about 30 dB at 15,000 Hz and 36 dB at 20,000 Hz. Yet Curve EH in Fig. 2 indicates that a treble equalization curve with only 10 dB of boost at 15,000 Hz is needed. The seeming paradox is explained by the fact that treble boost is not required to fully make up for treble losses. Only enough treble boost is needed to achieve flat response. Putting it differently, the rising response of the playback head (the portion of AD above 800 Hz in Fig. 1) compensates for a substantial part of the treble losses. Only the remainder of the treble losses must be made up by Curve EH in order to achieve flat response in the treble range. Thus we note that rising response of the playback head has a key role in treble compensation.

Where should equalization circuits be placed in the tape machine? One might expect that they could be placed in the record amplifier, or in the playback amplifier, or in a combination of the two. However, not just any combination will do, because some offer better results in terms of noise and distortion, while others offer worse.

Without yet explaining anything, one may offer a descriptive general rule: *Playback losses are equalized in playback and record losses in recording.* Thus Curve GF in Fig. 2 would be supplied by an equalization circuit in the playback amplifier. And curve EH would be supplied by an equalization circuit in the record amplifier.

Why this general rule? If the large amount of needed bass boost were supplied in recording, this would tend to apply excessive signal (magnetic field) to the tape and overload it, resulting in excessive distortion. Alternatively, one would have to greatly lower the recording level, resulting in a poor S/N ratio. Therefore bass boost is applied (largely or altogether) in playback. Moreover, Curve GF in Fig. 2 may be viewed in the guise of a treble cut characteristic. In this vein it serves to reduce noise of the entire tape recording system when used in playback.

Turning to treble boost, we must consider that noise, while prevalent at equal *amplitude* throughout the audio spectrum, is usually most evident from about 3,000 Hz upward. This is partly because of the human ear's sensitivity in the vicinity of 3,000 to 5,000 Hz. Mainly it is because of the increasing amount of random noise energy as one goes up each octave of the audio spectrum; the more frequencies per octave, the more must be the noise energy. (Clearly there are more frequencies between, say, 3,000 and 6,000 Hz than in the preceding octave of 1,500 to 3,000 Hz.) Thus we characterize tape and amplifier noise as high-pitched (hiss, spitting, frying, etc.) even though low-pitched noise is also present. If treble boost were applied in playback, it would magnify tape noise and noise of the record and playback amplifiers, resulting in a poorer S/N ratio. Therefore treble boost is applied instead (largely or altogether) in recording, where it only magnifies noise of the record amplifier.

A logical and important objection is in order at this point: Won't the treble boost in recording overload the tape (much as a large amount of bass boost might)? The answer would be yes if, for typical sounds, all frequencies had equal peak amplitudes. But for most recorded sounds desired by humans, particularly music, amplitude is usually a good deal less at high frequencies than at mid-range ones, as suggested in Fig. 3. This figure shows for a typical orchestral selection the relative peak levels of audio energy throughout the spectrum.

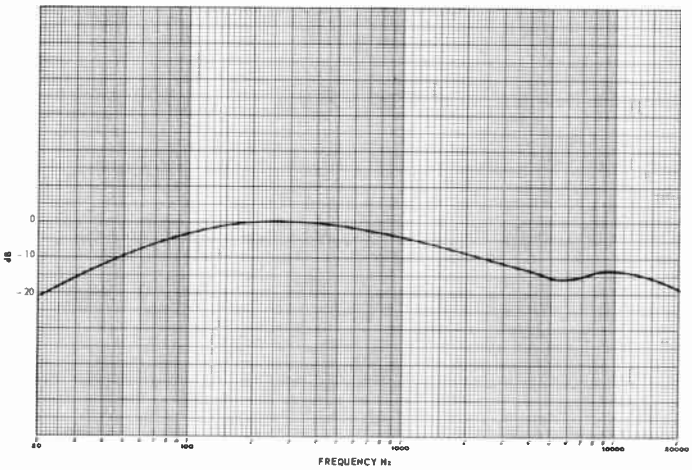


Fig. 3—Smoothed approximation of relative peak amplitudes for a typical orchestral selection.

Compared with peak amplitude at 400 Hz, there is a dropoff of 15 dB or more at higher frequencies.

Therefore, in dealing with the kind of sound generally recorded, a good deal of treble boost is feasible in recording. This boost is offset by the decline in amplitude of the higher frequencies, which helps prevent the tape from being overloaded. (Another preventative, when necessary, is reduction of recording level by the user.)

A Modified Pattern of Equalization

We have already noted that low noise is one of the three major goals of high fidelity. Put differently, we are interested in high S/N (signal-to-noise ratio). This can be achieved by recording more signal on the tape, especially at high frequencies, where the extra signal can mask the noise.

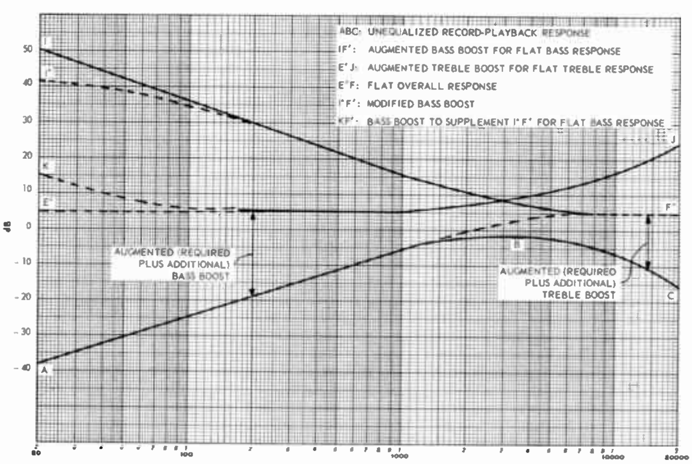


Fig. 4—A modified pattern of equalization to obtain improved signal-to-noise ratio (at 7½ ips).

Figure 4 shows a modified pattern of equalization that uses additional boost in recording yet results in flat response. Desired flat response is denoted by Line E'F'. It is 6 dB higher than the corresponding line EF in Fig. 2, reflecting an improvement of 6 dB in S/N ultimately achieved at higher frequencies.

Curve ABC, as before, is the unequalized record-playback response. The interval between AF' and ABC is the treble boost needed to approach flat response at the higher frequencies; that is, response 3 dB below flat at about 3,200 Hz, and increasingly flat as frequency rises. We may refer to this interval as "augmented" treble boost, consisting of the amount originally required in Fig. 2 plus an additional amount for higher S/N. (The interval between BC and the O dB line is the originally required treble boost, so that the remainder of the interval between AF' and ABC is the additional boost.) Thus Curve E'J is the augmented treble equalization needed for flat treble response. (E'J is the same distance from E'F' as ABC is from AF'.)

The required bass boost is the interval between flat response E'F' and drooping response AF' (keep in mind that AF' represents the response of an ideal head to a recording made with treble boost E'J). We may refer to this interval as "augmented" bass boost because it consists of the amount originally required in Fig. 2 plus an additional amount. (The interval between AB and the O dB line is the originally required bass boost, so that the remainder of the interval between AF' and E'F' is the additional bass boost. Thus

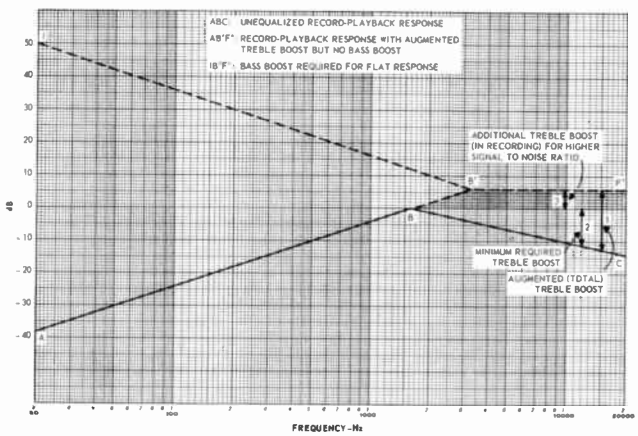


Fig. 4A—Basic scheme of the modified pattern of equalization.

Curve IF' is the augmented bass equalization needed for flat bass response. IF' is the same distance from E'F' as AF' is from E'F'.)

The story told by Fig. 4 is somewhat complex. It can be made more clear by presenting its essentials in simpler form in Fig. 4-A. (We haven't yet finished with Fig. 4 and shall return to it shortly.) ABC in Fig. 4-A represents in linear form the unequalized record-playback response. Line AB'F' shows the record-playback response that would result if there were only augmented treble boost and no bass boost. Augmented treble boost is depicted by Arrow 1, minimum required treble boost by Arrow 2, and additional treble boost by Arrow 3. Additional boost is further spelled out by the shaded area between BB'F' and BC. Given record-playback response AB'F', it remains to supply bass boost IB'F' in order to achieve overall flat response. AB'F' in Fig. 4-A corresponds to AF' in Fig. 4; and IB'F' in Fig. 4-A corresponds to IF' in Fig. 4.

Returning to Fig. 4, we may ask: Why stop at a 6 dB improvement in S/N at the higher frequencies? Why not add yet more treble boost in recording to achieve still higher S/N. The answer lies in Fig. 3, which shows that, relative to 400 Hz, the higher frequencies are down roughly 15 dB at 15,000 Hz and a bit more at 20,000 Hz. Correspondingly, this allows treble boost of about 15 dB at 15,000 Hz and a bit more at 20,000 Hz without excessive risk of serious tape distortion. Treble boost curve E'J does just about that, with no margin of safety to spare. In other words, in the present state of the art, treble boost in recording which approximates E'J is about as far as one dare go without risking excessive distortion in the upper end of the treble range. (Here lies the reason why some recordists still prefer 15 ips. Treble losses in recording are less than at 7½ ips, so that less treble boost is needed in recording and there is less risk of overloading the tape due to such boost. The recordist speaks of the greater "headroom"—margin between the amount of treble signal applied to the tape and the amount which causes tape saturation—available at 15 ips.)

It is difficult in practice for a tape amplifier to fully supply the amount of bass boost indicated by Curve IF' in Fig. 4. For one thing, an enormous amount of amplification is needed to achieve bass boost which at 20 Hz is up 44 dB from the reference line E'F' and still rising. High amplification is costly, may unduly magnify hum frequencies, and entails the risk of oscillation owing to phase shift or stray feedback. Therefore a preferred course is to allow bass boost to level off, as shown by I'F' in Fig. 4. Now bass boost is up about 35.5 dB at 20 Hz and soon reaches a maximum of 36 dB below 20 Hz.

To compensate for the levelling off of bass boost in playback, some bass boost may be introduced in recording, as shown be Curve KF' in Fig. 4. However, this isn't always necessary, because at low frequencies the playback head often tends to exhibit a slight rise in response owing to what is called the contour effect. Low frequencies correspond to long wavelengths (bar magnets) on the tape. In the presence of long wavelengths, the entire playback head, not only its gap, tends to react to the magnetic flux of the tape. The resultant rise in bass response may approximate KF' well enough to obviate the need for bass boost in the record amplifier.

If bass boost is supplied in recording because the contour effect is minimal, ordinarily this raises no problem of overloading the tape. Figure 3 shows that the typical decline of peak amplitudes at low frequencies would easily offset the bass boost of Curve KF'.

A final and key note on Fig. 4: Comparing this with Fig. 2, we observe that an equalization pattern which calls for increased treble boost in turn requires greater bass boost. Conversely, a decrease in treble boost is accompanied by a decrease in required bass boost. *In sum: A given recording characteristic implies a complementary playback characteristic; or a given playback characteristic implies a complementary recording characteristic.*

NAB Standard Equalization

The approach of Fig. 4 is followed by the NAB standards for tape recorder equalization. In fact, I'F' is the NAB playback equalization *characteristic* for 7½ ips (and for 15 ips as well). For greater clarity, I'F' is repeated in Fig. 5. We underscore the word "characteristic" because the NAB standard does not merely describe the frequency response of an equalization circuit in the playback amplifier (as is the case for RIAA phono playback equalization and for FM tuner equalization). True, measured frequency response of the tape

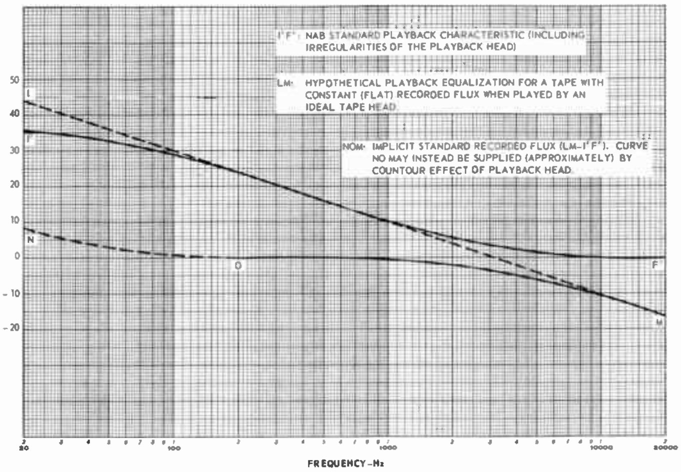


Fig. 5—NAB standard playback equalization and recorded flux.

playback amplifier is ordinarily quite close to Curve I'F', but it is not necessarily the same as I'F' in order to achieve flat response. The NAB playback characteristic in Fig. 5 is the *sum* of playback equalization provided by the tape amplifier plus irregularities in frequency response of the playback head. As already discussed, these irregularities tend to consist of some boost in the low bass region and a slight dropoff in the high treble region.

To illustrate, assume that a playback head is significantly deficient in high treble; that is, its output rises with frequency but at less than the theoretical 6 dB per octave rate of an ideal head throughout the audio spectrum. Then, if flat response is to be maintained, the playback amplifier must supply enough treble boost to compensate for the head's deficiency. *Together*, the amplifier and playback head supply the NAB playback characteristic, Curve I'F' in Fig. 5 (or something like I'F' if the contour effect is pronounced).

Without elaborate test equipment, how is one to ascertain whether a tape machine (playback amplifier plus playback head) provides the NAB playback characteristic? The answer lies in a standard test tape. (This is supposed to have been available from NAB by now, but hasn't yet been released. In its place, the Ampex test tape is customarily used.) The test tape contains a series of audio signals recorded at such relative levels that a tape machine with the NAB playback characteristic will provide flat response when playing this tape. That is, a meter connected to the machine's output will read equal output level for all the test frequencies.

Accordingly the manufacturer of a tape machine designs the playback equalization circuit to yield flat response when playing the standard test tape. The equalization circuit allows for bass and/or treble irregularities of the playback head he uses—that is, departures of the actual head from the response of an ideal head (AD in Fig. 1).

Some machines include adjustments which enable the technician or the user to touch up playback equalization on the basis of a test tape. It is then merely necessary to connect a meter to the machine's output, play the test tape (after making sure heads are cleaned, demagnetized, and aligned for azimuth), and touch up the playback equalization for flattest response as indicated by the meter. In some machines the VU meter can serve this purpose.

Now, what about treble boost in recording? Does NAB specify a treble recording characteristic in terms of what gets on the tape? Or does it specify a given amount of treble boost in the record amplifier, such as Curve E'J in Fig. 4? The answer is that, at least *directly*, NAB does neither. What NAB specifies is that, after playback equalization has been adjusted for flat response when playing a test tape, the record equalization should be adjusted for flat record-playback response.

There are good reasons for NAB not specifying a specific treble boost curve in the record amplifier: (1) Treble losses in recording vary among types and brands of tape, so that the amount of required treble boost varies according to the tape which the manufacturer envisions will be used with his machine. (2) These losses vary according to the amount of bias the machine manufacturer elects to use. He tries to come fairly close to the bias which achieves minimum distortion. However, minimum-distortion bias may entail excessive treble loss, causing the manufacturer to reduce bias somewhat rather than increase treble boost in recording and thereby heighten the risk of overloading the tape. In other words, the manufacturer may find that the increased distortion resulting from a slight decrease in bias is not as bad as the increased distortion which would occur if more treble boost were used.

The NAB Recording Characteristic

Indirectly, NAB does stipulate a recording characteristic in terms of the relative levels of recorded flux on the tape at various frequencies in the audio range. To explain, let us repeat our dictum on recording/playback characteristics: "A given recording characteristic implies a complementary playback characteristic; or a given playback characteristic implies a complementary recording characteristic."

Hence the NAB recording characteristic is Curve NOM in Fig. 5. Given playback characteristic I'F' and flat response, then the flux recorded on the tape must vary with frequency in the manner portrayed by NOM.

NOM is derived as follows. We draw LM to show the hypothetical playback equalization required if the recording characteristic were flat and if an ideal playback head were used (or a playback head with its irregularities fully compensated in the playback amplifier). That is, LM declines 6 dB per octave with increasing frequency, thereby complementing the playback head's 6 dB rise per octave. However, any departure of the acutal playback characteristic (I'F') from LM implies a corresponding (complementary) departure of recorded flux from a flat characteristic. Thus NOM is the difference between hypothetical and actual equalization, namely between LM and I'F'.

At the low end, I'F' supplies less than the hypothetical bass boost. Therefore the deficiency must be made up in recording by bass boost NO (unless the deficiency is made up by the contour effect of the playback head). At the high end, I'F' does not drop as rapidly as LM; thus I'F' in effect is contributing treble boost. And a corresponding drop in treble must occur in recording, namely the treble decline of recording characteristic NOM.

The implied recording characteristic NOM—flux on the tape—is the sum of magnetic losses in recording, equalization in the record amplifier, and response irregularities of the record head. All told and together with playback characteristic I'F', they produce flat response.

If, after all, there is an NAB standard recording characteristic, why doesn't NAB specify this explicitly (in the way that the RIAA phono recording characteristic and the FM broadcast characteristic are specified)? The answer lies in the kind and quality of laboratory test equipment required in order to measure recorded flux. It is far easier for the manufacturer, technician, or user to check the playback characteristic with the aid of a test tape and meter than to measure recorded flux. Since a playback characteristic implies a matching recording characteristic when overall response is flat, then if the playback characteristic is known to meet the standard (on the basis of playing a test tape) the recording characteristic perforce also meets the standard.

This signifies that if my tape machine and yours both have flat record-playback response, and if both produce flat output when playing the standard test tape, mine will record tapes that play back flat on yours, and vice versa.

We have several times commented on NO in Fig. 5. To bring things together, at the cost of repetition, this slight bass boost may be achieved by an equalization circuit in the record amplifier, by the contour effect of the playback head, or by a combination of the two. Even if there is no appreciable contour effect, the manufacturer may choose not to supply bass boost in recording. This is consistent with the NAB standard for frequency response, which permits record-playback response to be down 3 dB at 50 Hz.

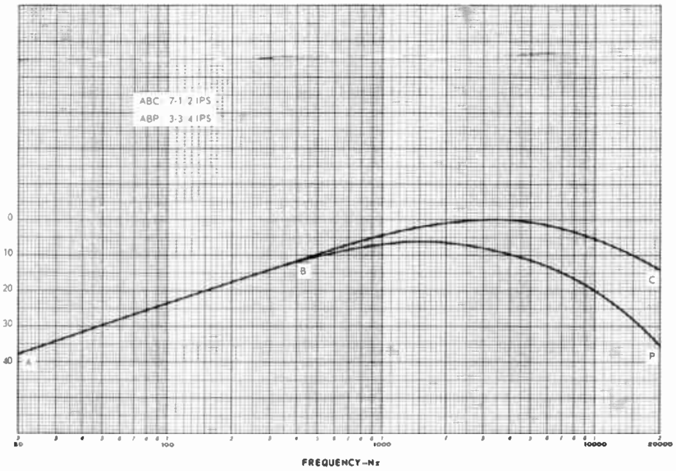


Fig. 6—Comparison of unequalized record-playback response at 7½ and 3¾ ips.

Other Speeds, Other Tapes, Other Matters

All that we have said about problems and techniques of tape equalization at 7½ ips also applies in essence to the lower (higher) speeds in home use. However, as speed is reduced, equalization must be changed, the principal reason being the greater magnetic losses that occur in recording at reduced tape speed. This is illustrated in Fig. 6, which compares unequalized record-playback response at two speeds, 7½ and 3¾ ips.

Why do recording losses increase with reduced speed? The answer lies in the fact that these losses actually depend on recorded wavelength rather than on frequency as such. We refer again to the length of the bar magnets that in effect are recorded on the tape. At a given frequency, a corresponding number of bar magnets are recorded per inch of tape. To illustrate, consider a 1,000 Hz tone, which is recorded as 2,000 bar magnets—one magnet for each positive portion of an audio cycle and one for each negative portion. At a tape speed of 7½ ips, the 2,000 magnets are recorded on 7½ inches of tape, so that the length of each magnet is 7.5/2,000 inches or .00375". But at 3¾ ips, the length is 3.75/2,000 inches or .001875"—half as long. Earlier we pointed out that the shorter the wavelength (bar magnet), the greater the loss due to self-demagnetization and bias erase. Thus at a given frequency the wavelength is reduced and the recording loss is increased as tape speed is reduced. In going, say, from 7½ to 3¾ ips, losses are of the same magnitude at 5,000 Hz as they were at 10,000 Hz at the higher speed; losses are of the same magnitude at 10,000 Hz as they were at 20,000 Hz; etc.

To offset the greater treble loss at low speed, more treble boost is needed in recording. But the requisite amount would

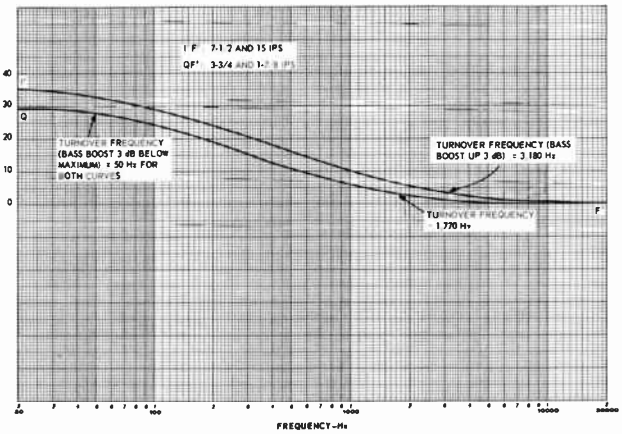


Fig. 7—NAB standard playback equalization curves (including playback head characteristics).

overload the tape and cause excessive distortion. A viable alternative is to reduce treble boost and match this with reduced bass boost in order to attain flat response. As pointed out in our dictum on record/playback characteristics (in conjunction with Figs. 2, 4, and 4-A), reduced treble boost correspondingly calls for reduced bass boost in order to achieve flat response.

In this light the NAB standard for 3¾ and 1⅞ ips calls for a playback characteristic with 5 dB less bass boost than the characteristic for 7½ ips (which is also the playback characteristic for 15 ips). Fig. 7 shows the two playback characteristics. Curve I'F' is the NAB standard characteristic for 7½ and 15 ips, and QF' is the standard for 3¾ and 1⅞ ips.

I'F' entails a total of 36 dB bass boost; QF', 31 dB. In technical terms, I'F' is described as having turnover frequencies of 3,180 and 50 Hz (or, respectively, time constants of 50 and 3,180 microseconds). This signifies that the curve has achieved 3 dB boost at 3,180 Hz, and at 50 Hz is 3 dB below maximum boost. QF' has turnover frequencies of 1,770 and 50 Hz (or, respectively, time constants of 90 and 3,180 microseconds), signifying 3 dB boost at 1,770 Hz, and boost 3 dB below maximum at 50 Hz. (The relationship between time constant t in microseconds and turnover frequency f in Hz is expressed by $t = 159{,}155/f$. The factor 159,155 derives from the technical relationship between the values of capacitance and resistance needed in an equalization circuit to achieve a 3 dB change in frequency response.)

According to Fig. 7, two playback equalization characteristics are used for four speeds, one characteristic for 7½ and 15 ips, and the other for 3-¾ and 1-⅞ ips. But the sense of this article is that each tape speed requires its own equalization in order to obtain optimum results with respect to frequency response, S/N, and distortion. In other words, why don't we have four standard equalization characteristics instead of two? The indicated answer is that four characteristics would complicate matters too much for those making tape machines, and perhaps for those using them. An equalization characteristic appropriate for a given speed can be used for the next higher speed without undue departure from optimum performance. So all in all, two playback characteristics for four speeds affords a practical compromise without unduly deleterious consequences.

What happens to equalization requirements as tape formulations change, resulting in higher output, lower noise, better treble, different bias requirements? So far as playback equalization is concerned, the answer in essence is *nothing*, or very little. Essentially it is only record equalization that changes (except in the unlikely event that changes in tape formulation would someday result in a new standard playback characteristic, which would be a very unhappy day for those with substantial collections of recorded tapes).

If a new tape has increased treble output, the tape machine manufacturer may choose not to extend treble response but to reduce the amount of treble boost supplied by the record amplifier, thereby lessening the chance of tape saturation. Alternatively, he can leave treble boost about the way it was, and increase bias to achieve less distortion, yet without reducing treble. Or, in increasing bias, his purpose may be not to reduce distortion but to permit recording at a higher level at the same distortion as before, thus improving S/N. Or the manufacturer may follow a compromise course which results in some combination of improvements in treble response, distortion, and S/N.

If a new tape has higher output, the machine manufacturer can in similar fashion take advantage of this to improve S/N, treble response, distortion, or a combination of them. Higher tape output enables the machine to apply somewhat less signal to the tape without sacrificing S/N; hence there is less danger of tape saturation, permitting more treble boost for better treble response; or permitting more treble boost together with more bias for lower distortion. Higher tape output enables the machine to reduce recording level for less tape distortion. Higher output enables the machine to keep bias and treble boost as before, with an increase in recording level to achieve higher S/N, yet without increase in distortion.

In similar ways, reduction in tape noise not only permits higher S/N but can also be translated into improvements in distortion and treble response, which might entail changes in treble boost, bias, or recording level. Changes in a tape's bias requirements do not in themselves entail changes in treble equalization. However, changes in bias requirements for a tape tend to accompany changes in noise, treble, and output characteristics of the tape, and it is then that the machine manufacturer may find it advisable to change record equalization.

Does use of the Dolby noise reduction system (or other noise reduction systems such as the Burwen) affect tape equalization? The answer, essentially, is only in recording if at all. Introduction of the Dolby system in a tape recorder installation does not necessarily call for a change in recording equalization. One records as before, except than the incoming signal first goes through a treble-boosting Dolby circuit; and one plays as before, except that the playback signal afterward goes through a matching treble-reducing Dolby circuit to achieve noise reduction and restoration of flat response. Overall record-playback response is not changed by Dolby, which therefore does not *compel* a change in tape equalization.

On the other hand, the Dolby (or a similar) system may *invite* changes in equalization for reasons already suggested here. To illustrate, assume that Dolby achieves a 10 dB reduction in noise. The tape machine manufacturer might decide to sacrifice part of this improvement, say 4 dB, in exchange for better treble response. He could adjust the tape machine (adjust the reading of the record-level indicator) to operate at 4 dB lower recording level, permitting 4 dB more treble boost without increase in danger of tape saturation. Or he could exchange part of the Dolby reduction in noise for improvement in distortion, by using more bias along with more treble boost, as well as a lower recording level. Or he could achieve some improvement in all three respects—noise, treble response, and distortion. Æ

tape

I suppose few products in the Hi Fi world (with the exception of loudspeakers) are advertised with such exuberance as magnetic tape. Each manufacturer proclaims that his tape is the absolute best with the lowest noise, highest output and widest frequency response. New coating materials and improved formulations are launched almost every month with fanfares of publicity. And so we have Low Noise, Ultra-Dynamic, Super Dynamic, High Efficiency, Low-Noise-High Output and dozens of other permutations—no wonder many people are confused and a little skeptical!

Tape recording parameters are very much of a compromise with bias, equalization and output adjusted for frequency response, noise and distortion—all conflicting requirements. For example, a high value of bias might well reduce distortion at mid-frequencies but at the cost of a lower output above 10 kHz or so. Thus a tape with a rising frequency response, a characteristic of the new formulations, will help matters considerably. Figure 1 shows what happens when a Low Noise tape is played on a recorder matched to a normal tape. It will be seen that the response rises from about 3 kHz. If the amplifier treble control is turned down to get a more or less flat response, the inherent noise will be reduced too. Present day recorders are fitted with a switch—usually labeled Normal and High to cope with these LN tapes by changing the bias and equalization and the net result is a reduction of noise by some 4 dB—plus an extended high frequency response. Machines are matched for one particular tape at the factory and the first objective of our survey was to determine how the tape characteristics varied. These tests were made with a TEAC 1230, a medium priced deck, selected for its high overall performance and because of the accessibility of the bias adjustments. First, it was carefully matched for Scotch 207 tape so the response was within 0.5 dB up to 20 kHz and then the other tapes were checked, without changing the bias. Frequency runs were made from 1 kHz to 20 kHz with a recording level of—20 VU. Levels were adjusted to 0 dB at 1 kHz and the results tabulated for three frequencies, 10 kHz, 15 kHz and 20 kHz as shown in the chart. Variations at 15 kHz ranged from -4.5 dB for the Dak to +3.1 dB for the Maxell UD. In theory, tapes with the highest increase would have the best signal-to-noise potential but this is also dependent on sensitivity and other factors.

For the next series of tests, the recorder was adjusted to suit each individual tape, using only the bias control and frequency responses taken at the standard input level of -20 VU. Equalization was unchanged and only the bias was adjusted—not too difficult to do with most recorders. Results are shown in the graphs. The response below 1 kHz is not shown as the low frequency deviations were insignificant. Then the measurements were repeated at a higher signal level—0 VU, a drastic test which shows the effect of tape saturation or maximum capacity. This has long been a limiting factor on obtaining good transient reproduction and it is also why it is still standard practice to measure recorders at the 20 VU level. Obviously this cannot give a true indication of performance and in the future our tests of recorders will be made at the higher signal level. It will be noticed that all the tapes show some falling off above 15 kHz but even the worst has more 'headroom' than the best tapes of a few years ago.

Distortion was measured at 0 VU and +3 VU at 1 kHz and signal-to-noise computed. The standard 3% distortion level was used and the ASA weighting employed. It was felt that this method gives a truer indication as the curves are based on human hearing characteristics and it also reduces the effects of low frequency preamplifier noise. (See figure 2). The figures *do* include some residual electronic noise —note also that a half-track machine would give a further 3 dB improvement. Finally, output at 0 VU input (1 kHz) was measured using 0 dB with the Scotch 207 as a reference.

Dropouts

Dropouts, not the scholastic kind, can be caused by minute imperfections in the tape which push it away from the

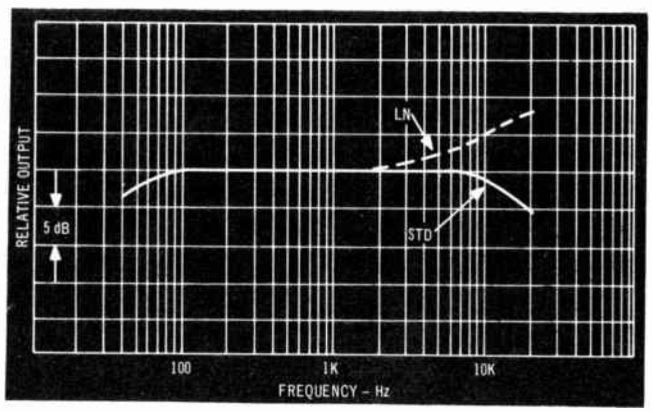
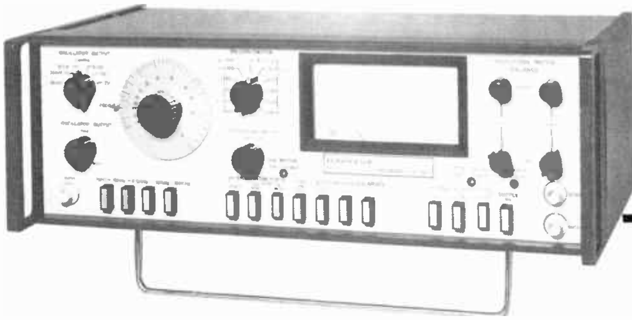


Fig. 1—Showing the frequency response of a typical LN tape played on a recorder adjusted for normal tape.

survey



head. The effects are more audible at high frequencies and tape speed plays a part too. A dropout might pass unnoticed at 7½ ips but sound horrendous at 1⅞ ips. And so they are more noticeable on cassette machines where mechanical difficulties compound the problem.

The usual method of assessing dropouts is to record a high-frequency signal on the tape for 3 to 15 minutes, then playing back and watching the variations on an output meter. This is tedious and seems a little crude and so work is proceeding on an automatic counter detector (we are just lazy, that's all). In the meantime, the 15 tapes were checked using the first method and all were found satisfactory.

As far as open reel is concerned, we had long thought the dropout problem (when it was not due to pressure pads) was due to bad handling and some tests made by the Boston Audio Society seem to confirm this opinion. These tests were made by Alvin Foster who writes in the BAS publication. "Dropout is particularly severe in the first and last feet of most open reel tapes due mainly to the mistreatment that these sections endure in the normal loading and unloading of the tape. This is the reason why some manufacturers include a few feet of leader tape and why tape recording perfectionists never record a signal on the first or last few feet of tape. . . . I was surprised to find such noticeable deterioration between new and slightly used tapes and was convinced that careful handling of tape is a prerequisite to good recording. Even the type of reel one used affected dropout. Take up reels which cover the tape with only one or two small holes for threading protect it because the larger plastic area covering the tape inhibits pinching or creasing of tape as the reel is handled. The pressures exerted when handling the reel of tape are distributed more evenly over the tape edges."

We would add that the back coatings used by Scotch and Capitol are definitely a step in the right direction as they not only tend to improve traction but strengthen the tape as well.

What *do* the tests show? In brief, they confirm that the new generation of tapes, Low-Noise, Super-Dynamic or whatever, do give a worthwhile improvement; moreover, signal-to-noise is so good that you can record at a lower level and avoid the risk of peak overloading. But the divergencies mean that the recorder must be adjusted to suit. That is, if you want the best results—and who doesn't.

Next month, we compare cassette tapes.

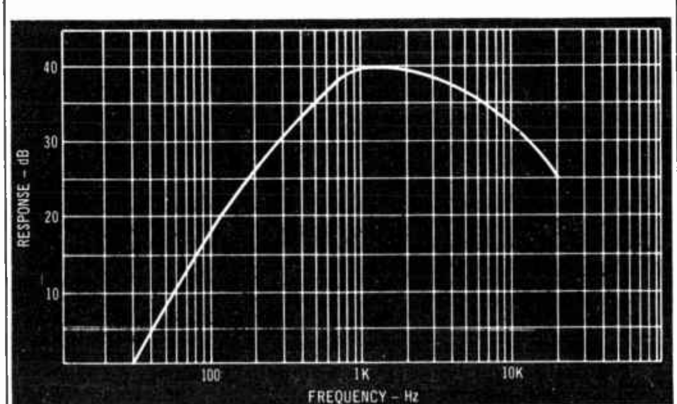
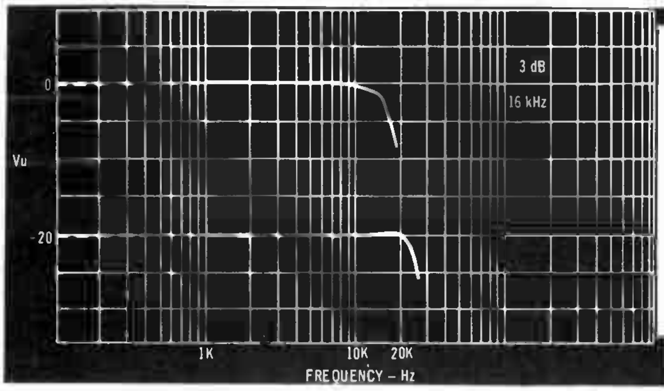


Fig. 2—ASA weighting characteristics.

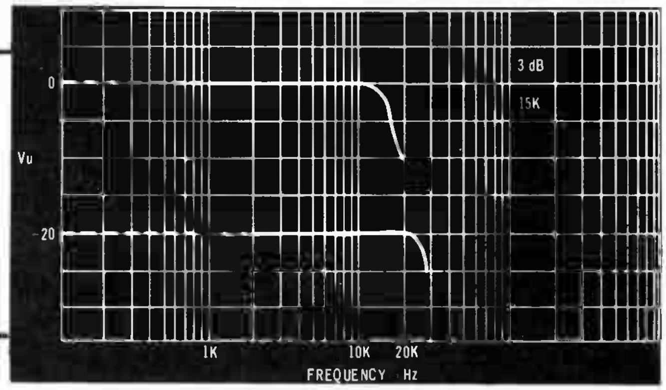
TAPE	10 kHz	15 kHz	20 kHz
Agfa P36	0	0	-1
Ampex LNP	+0.5	+0.3	0
BASF LP 35	+1.8	+2.0	+2.0
Capitol 2	+2.0	+3.0	+3.0
Dak LNE	-2.0	-4.5	-7.0
Irish 274	+1.5	+1.8	+1.8
Maxell LNE35-7	+1.7	+1.7	+1.0
Maxell UD50-7	+2.5	+3.1	+3.6
Memorex	+1.0	0	-0.5
Scotch 207	**0**	**0**	**+0.5**
Scotch 228	+1.0	+0.3	+0.5
Sony SLH 180	+2.5	+2.5	+2.0
TDK 7	0	-1.0	-1.6
TDK SD	+2.0	+2.2	+2.3
Tracs Plus	0	-1.0	-2.0

Table 1—Frequency responses compared with Scotch 207



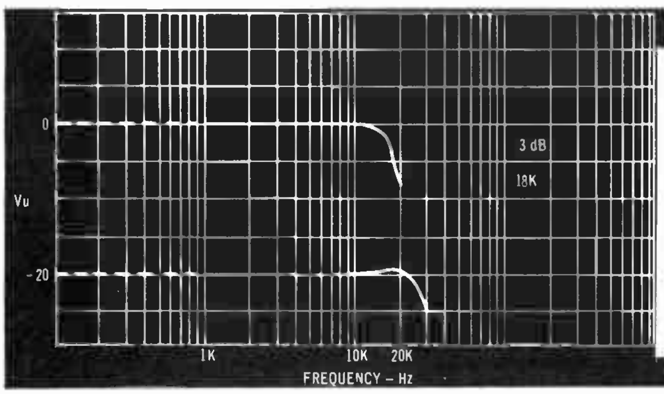
Agfa PE 36

1 mil polyester, plastic case, leader tape. Output +1 dB, signal/noise 67 dB. Distortion 0 VU 1.1%, +3 VU 2.5%.



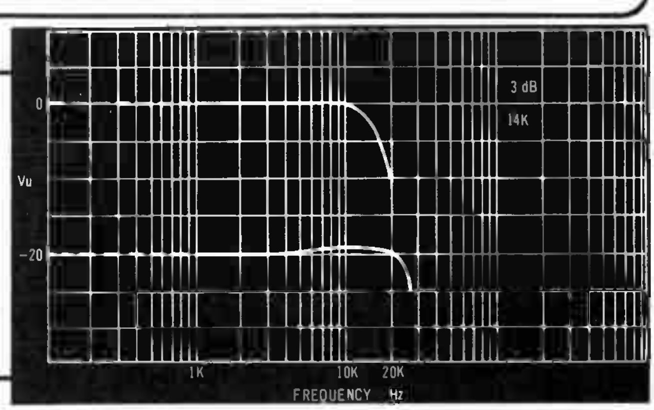
Ampex LNP

1 mil polyester, plastic case, no leader tape. Output 0 dB, signal/noise 67 dB. Distortion 0 VU 0.95%, +3 VU 2.8%. Frequency response nearly identical to Scotch 207.



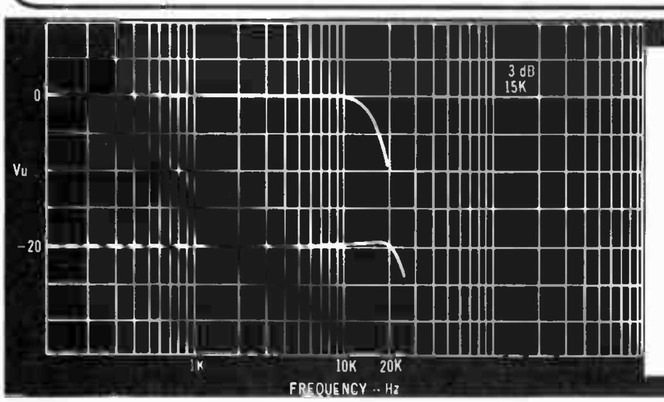
BASF LP 35

1 mil polyester, plastic case, leader tape. Output +0.5 dB, signal/noise 69.5 dB. Distortion 0 VU 0.6%, +3 VU 1.1%. Low distortion with good headroom.



Capitol 2

1.5 mil polyester, plastic case, no leader tape. Output 0 dB, signal/noise 68 dB. Distortion 0 VU 0.75%, +3 VU 2.0%. "Cushion-aire" back coating.



Dak Low Noise

1 mil polyester, cardboard box, no leader tape. Output + 0.5 dB, signal/noise 62 dB. Distortion 0 VU 1.2%, 0 VU 3.2%. An inexpensive tape with excellent frequency response, good headroom but with slightly inferior signal/noise.

If you're looking for a great buy in a TEAC 450...

The only cassette deck in the world that can equal the precedent-shattering 0.07% WRMS record and playback wow and flutter of the TEAC 450 is the TEAC 360S. In fact, their performance specs are identical.

They differ only in breadth of features. And therefore in price. So if you don't *need* all the sophisticated mic-line mixing applications of the 450, don't get it.

Get the 360S.

It has slide controls with mic/line inputs, a memory digital counter, a tape flow indicator, a peak level Light Emitting Diode, a removable head cover and 3-position bias and equalization switches. Not to mention Dolby* circuitry enhanced by 8 calibration controls and a Dolby tone generator. Quite a package.

Its incredible 0.07% wow and flutter makes the 360S a good deal better than anyone else's cassette deck. And its price makes it a good deal—better than our own 450.

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Donald R. Hicke

BUILD YOUR OWN Tape eraser

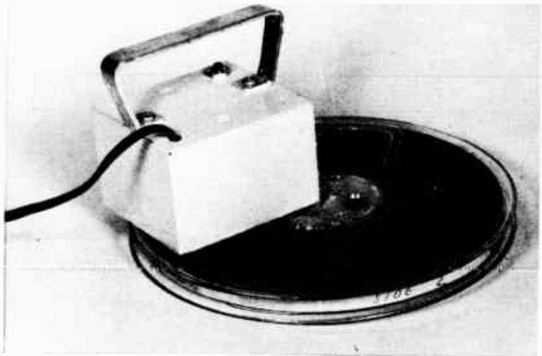

Fig. 1

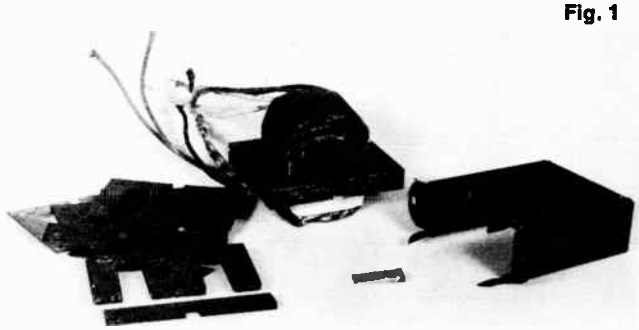

Fig. 2

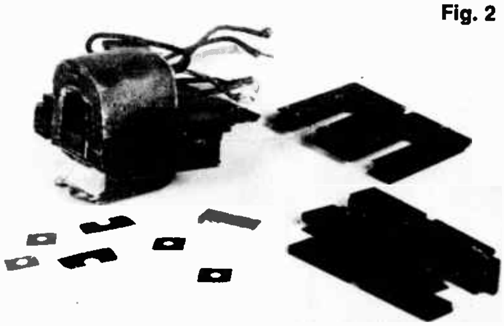

Fig. 3

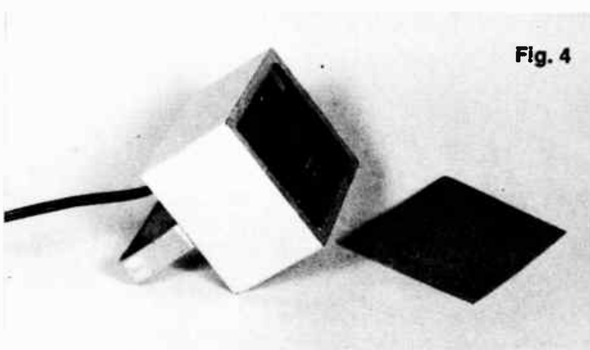

Fig. 4

THE NEXT best thing to using a fresh, new roll of tape for recording that important concert or other one-of-a-kind performances is to bulk erase an old reel of tape. This is especially important if the erase head on your recorder is not very efficient, or if you use high-output tape, which is difficult to erase. Otherwise, the previously-recorded material may be audible during the more quiet passages of the new recording.

A bulk tape eraser consists of a coil of wire wound around a laminated-iron frame open on one side. When the coil is connected to an a.c. source and held near a reel of tape, the magnetic circuit is completed through the iron oxide coating on the tape, effectively erasing any recorded material. Commercial tape erasers use a specially-designed coil, but you can make your own eraser using almost any old transformer. Good results have been obtained with filament transformers, audio output transformers, and power-supply chokes. All you have to do is take the transformer apart and rearrange the core pieces. The accompanying pictures will show you how.

To use the eraser, hold it directly against the reel of tape and turn on the power. I used a push-push switch in this version, but this is only a convenience item. You will hear and feel the tape vibrate at 60 Hz. Move the eraser all around the reel, then repeat on the other side. Slowly move the eraser several feet away from the tape before turning off the power. ●

Fig. 1—This transformer was salvaged from a vacuum-tube amplifier. Pry up and straighten the four tabs, and remove the frame. Pull out the wooden wedge(s), and remove the core pieces by prying them apart with a sturdy pocket knife. Do not cut into the core winding! **Fig. 2**—Reassemble the E-shaped pieces, facing them all the same way. Discard the I-shaped pieces. Clip off and discard any extra-length tabs on the two outside core pieces. **Fig. 3**—Reinsert the wedge(s), and clip off all wires except the primaries (plate-to-plate for an audio output transformer). **Fig. 4**—Pass the a.c. connection cord through a rubber strain-relief grommet and then through a hole in a suitable box. Connect the a.c. cord to the coil, making certain each connection is well taped and the whole is well insulated. Locate the grommet, and mount the coil. Cover the bottom of the box with cardboard. Fasten a handle with screws, and paint the box.

Dolby B-Type Noise Reduction System

Robert Berkovitz and Kenneth Gundry*

An unsatisfactory signal-to-noise ratio has remained the major obstacle to attaining an adequate level of performance from consumer media for music reproduction. This is especially true of the music-cassette, because of its slow tape speed and narrow track width, but it is also true of stereo FM broadcasting and the phonograph record. Although hopes were raised in recent years that further development of magnetic tape would eliminate its inherent noise as a problem, these hopes have been frustrated by the relatively modest gains achieved, and by studies which indicate that the available signal-to-noise ratio of present-day tapes is very near the maximum value imposed by theory.

It is therefore not surprising that numerous attempts have been made to devise methods of noise reduction satisfactory for professional and consumer use. However, almost all of the methods proposed have had unacceptable drawbacks.

The effectiveness of single-ended (non-complementary) systems, for example, which are designed to be used only during playback, extends only as far as the listener's willingness to sacrifice musical information. In principle, all playback-only systems depend upon the idea that the signal and the objectionable noise occupy separate domains; if this is correct, then the problem of noise reduction is one of defining the boundary between the domains, in terms of frequency and/or level, and designing a circuit to suppress everything on the "noise" side of the boundary. However, if the noise spectrum of ferric oxide cassette tape is taken as an example (see Fig. 1), it is seen that the noise, when passed through a standard DIN weighting network simulating the ear's sensitivity, remains considerable in the 1-4 kHz range. Since this range includes many of the lower harmonics and upper fundamental tones in music, it is not possible to suppress it, even at low listening levels, without serious loss of information. On the other hand, the noise within this range is so disturbing that if it is not reduced by such a circuit, the amount of subjective improvement obtained is minimal.

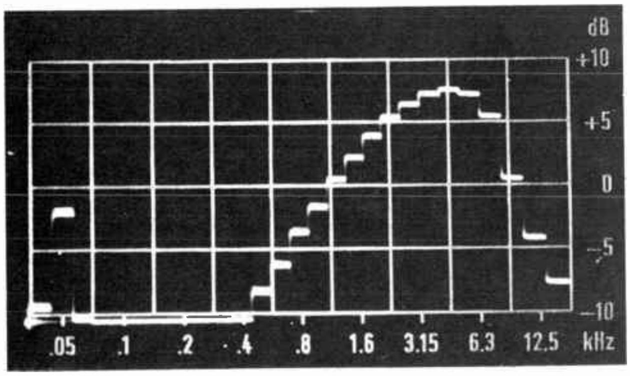

Fig. 1—DIN-weighted noise spectrum of low-noise ferric oxide cassette tape displayed on screen of real-time analyzer. Vertical scale is relative only; DIN noise measurement of sample shown was —47 dB with respect to 200 nWb/m. A substantial amount of the noise present falls well within the range of musical fundamental tones.

Complementary methods, i.e., those which require some signal processing or encoding during both recording and playback, offer greater promise, but can also present difficulties when put into practice. Pre- and de-emphasis schemes, for example, in which high frequencies are increased during recording and decreased by the same amount during playback, are only of limited value. Even in FM broadcasting, where such standardized pre-emphasis has been employed for many years, the usefulness of its continued application is in doubt. The primary problem is that modern microphones and recording equipment now routinely reproduce high frequencies at amplitudes so high that they were considered unlikely when current FM standards were set. Broadcasters are now forced to use limiters to prevent overmodulation, if they also wish to maintain reasonable levels at middle and low frequencies. In magnetic tape recording, pre-emphasis is difficult to use because tape saturation occurs at lower levels at high frequencies. Since high-frequency signals already present problems in cassette recording because of their short wavelength, added pre-emphasis would complicate a task which is already difficult.

The compandor type of noise reduction system, in which the dynamic range of the signal is compressed during recording and expanded again during playback, offers more promise. However, a simple compandor, even if precise in its action, also presents problems. In recordings and broadcasting, one of the most serious drawbacks of the compandor is the danger of signal overshoot, which can result in distortion or overmodulation of a transmitter.

An even more serious problem of compandors, from the listener's point of view, is noise modulation. When a conventional full-band compandor is used, low-level passages are recorded at a level higher than normal. They are then played back at reduced level, restoring correct signal dynamics and reducing noise at the same time (see Fig. 2). There can be no noise reduction effect during high-level passages, because this would require increasing the level of such passages during recording, resulting in overload. The simple compandor therefore requires that one assume that noise is not objectionable when the signal level is high. However, this is not always the case. A high-level bass drum beat, for example, does not mask high-frequency tape hiss; as a result the drum and other instruments introduce noise modulation during playback—each note is accompanied by a "swish" as the noise level rises for the duration of the note. While it is not audible with all types of program material, noise modulation limits the usefulness of the compandor considerably.

The extreme diversity of available source material and the high quality of present-day master recordings are the factors which really determine the conditions to be met by a satisfactory noise reduction system for home use. It must be remembered that many home listeners own playback equipment with very low distortion and wide frequency range,

*Dolby Laboratories, Inc.

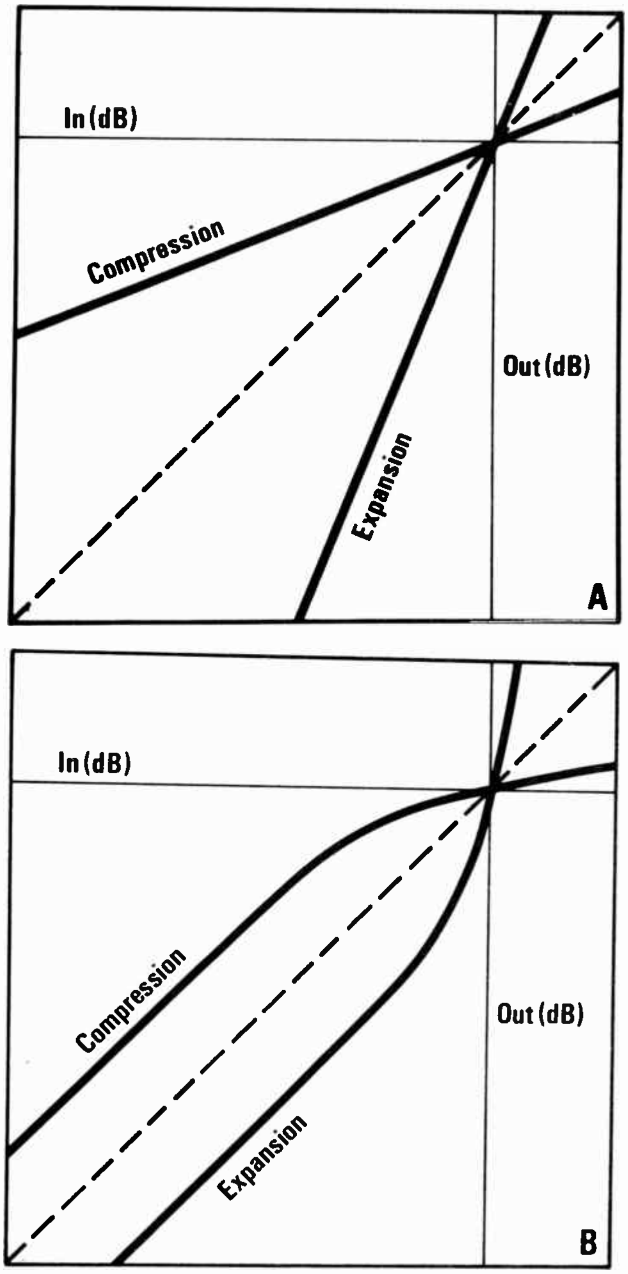

Fig. 2—Transfer characteristics of two conventional compandors (solid lines). A, constant-slope type; B, high-level type. Since compression and expansion are functions of signal amplitude only, in a single frequency band, such compandors fail to suppress noise whenever natural masking fails (see text).

disclosing audible effects which might have passed unnoticed in earlier times. Therefore, it is especially important that the program be recovered accurately after noise reduction, without addition of any audible sound. For the listener's sake accuracy of recovery and effectiveness of the system should not require adjustment of system parameters to match various kinds of program material. At the same time, the size and cost of the system should introduce no obstacle to its use. Furthermore, as a practical matter for the industry, it is clear that the system should require no modification of present professional practice in master recording, duplicating, or broadcasting.

The Dolby B-Type Noise Reduction System

The Dolby B-Type circuit is a specialized form of compandor which avoids the usual deficiencies of compandors. The operational principle of the B-Type system is complementary low-level compression and expansion in a frequency range which varies in bandwidth as the signal changes.

Most objectionable noise encountered in home listening is at middle and high frequencies, from about 500 Hz to the upper limit of audibility. In the interest of circuit economy, the action of the B-Type circuit has therefore been limited to this range. A feedback control circuit adjusts system parameters automatically as a function of signal level and spectrum, so that the system's action complements the psychoacoustic masking of noise which occurs naturally in the course of the program. A block diagram of a Dolby type of noise reduction system is shown in Fig. 3. The circuits used for encoding (during recording or transmission) and decoding (during playback or reception) are quite similar and can be considered as the same circuit, switched to operate in either mode.

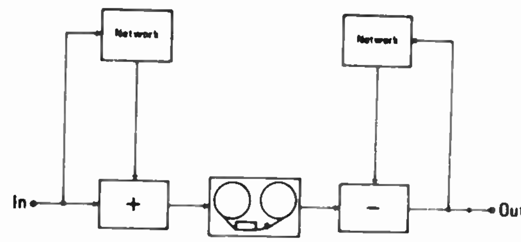

Fig. 3—Block diagram of Dolby type noise reduction circuit as used in typical record-reproduce chain.

The compression and expansion characteristics of the Dolby B-System are fixed and are referred to Dolby Level, a specific internationally standardized reference level. In the case of cassette tape, Dolby Level is a flux of 200 nWb/m; in FM broadcasting, Dolby Level is ± 37.5 kHz deviation.

Figure 4 is a block diagram of a switchable (encode-decode) B-Type circuit. There are two paths which the input signal follows: a *main path* (at the lower part of the figure) in which no change other than linear amplification occurs, and a *secondary path*, a variable filter through which only low-level, high frequency components of the input signal are allowed to pass. To encode the signal, the output of the secondary path is combined with signal in the main path *additively;* this boosts low-level, high frequency portions of the signal. Decoding is accomplished by feeding the secondary path from the circuit output, which is opposite in phase to the input (note phase inverter in Fig. 4); the secondary path is then part of an a.c. negative feedback loop which reduces output, i.e., the output of the secondary path is combined with the main path *subtractively.* In the decode mode, therefore, the circuit reduces the level of precisely the same information which was increased in level during encoding.

As Fig. 4 indicates, the action of the B-Type circuit is controlled by the output of the filter in the secondary path. Above a fixed threshold level, the bandpass of the filter, in turn, is modified by the d.c. feedback loop.

At very low levels, i.e., below the threshold, which at high frequencies is about 40 dB below Dolby Level, the output of the filter is not sufficient to generate d.c. feedback; consequently, the output of the secondary path is simply proportional to signal level within the filter pass band. The output of the circuit is then essentially as shown in Fig. 5.

As signal level rises above the threshold level, the rectified filter output is returned to the FET gate where it is applied as negative feedback, raising the filter cutoff frequency so

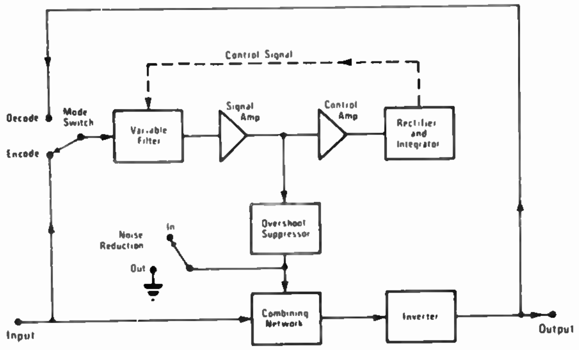

Fig. 4—Block diagram of Dolby B-Type noise reduction circuit. The configuration shown can be switched to encode or decode the signal.

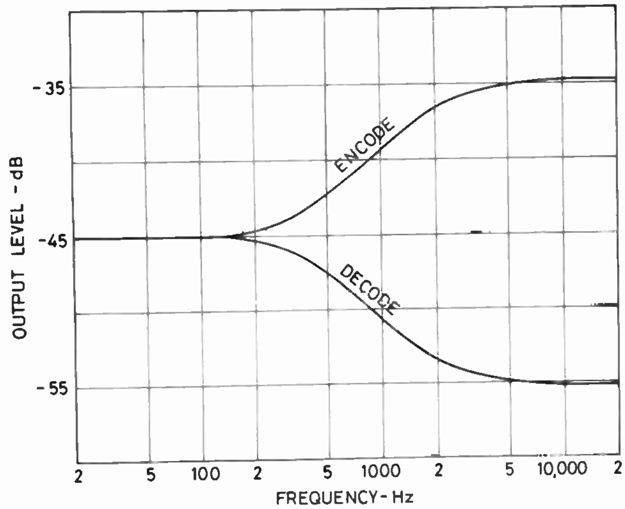

Fig. 5—Output of B-Type encoder and decoder circuits under low-level input signal conditions. The two operations are symmetrical and the result is an overall frequency response which is level.

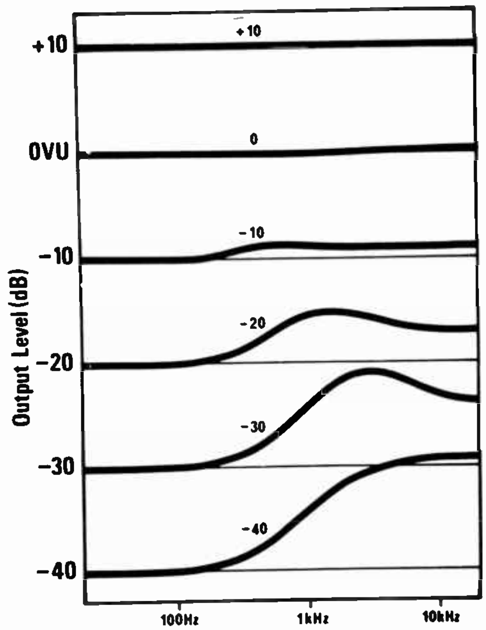

Fig. 6—Characteristics of encoding processor at several levels. The gradual reduction in boost with increasing level avoids possible tape overload.

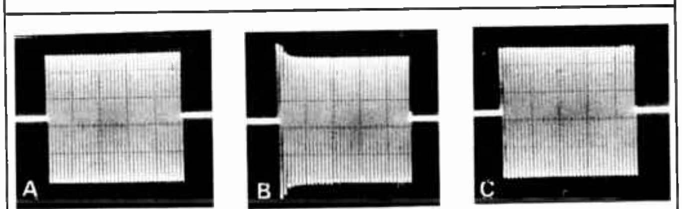

Fig. 7—Effect of the B-Type circuit on a tone burst; frequency = 3 kHz; burst duration, 12 milliseconds; low level = 40 dB; high level = +6 dB; (A) Input to system; (B) Encoded, (C) Encoded and Decoded.

that the output of the secondary path, while still increasing, no longer does so in proportion to the change in signal level. As signal level becomes even larger, the increasing d.c. feedback generated restricts the filter bandwidth further, and near Dolby Level the output of the secondary path remains relatively constant. The net effect is that the secondary path has no audible effect on output at low frequencies, and increasing effect with increasing frequency and decreasing level to about 40 dB below Dolby Level. At high levels, the effect of the extra signal is so small as to have no significance; at low levels, in the spectral region in which noise reduction is required, the increase during encoding is as much as 10 dB, and is of considerable importance.

The manner in which the secondary path changes from constant-gain to constant-output is determined by the adjustment of gain within the feedback loop. In addition, the exact variation in filter bandpass with changing level is set optimally by making the control amplifier frequency-dependent. The overall frequency response of a B-Type encoder circuit for different input levels is shown in Fig. 6.

A compandor operating over a wide frequency range must be designed to take into account the problem of noise modulation discussed above. If some high-level passages in the program differ sufficiently in frequency content from the noise components present, the latter will remain audible during the program in many cases. However, these passages cannot be increased in level when encoded, because of the danger of overmodulation. Under these conditions, compression may be applied intermittently, and high-frequency noise modulated audibly by mid-frequency components of the signal. The B-Type circuit overcomes this problem because it continues to function when a high-level signal occurs within its operating range; instead the feedback control shifts the range upward in frequency. This avoids the danger of overmodulation, but retains full noise reduction at frequencies higher than those masked by the signal.

The attack time of the B-Type circuit is dependent on the amount and rapidity of the signal change, due to the nonlinear design of the integrator, varying from about 100 milliseconds to as little as 1 millisecond. The recovery time of the rectifier-integrator is shorter than that of the human hearing system, about 100 milliseconds.

All compressors exhibit overshoot, including the B-Type circuit. However, the dual-path approach used makes it possible to reduce the amplitude of overshoots significantly. Overshoot, which can occur only in the secondary path (where it can be suppressed without affecting the main signal) is comparatively small, and essentially disappears when the signal is decoded again. When signal levels are low, or when changes in signal level take place slowly, there is no overshoot problem; when signal changes are large and rapid,

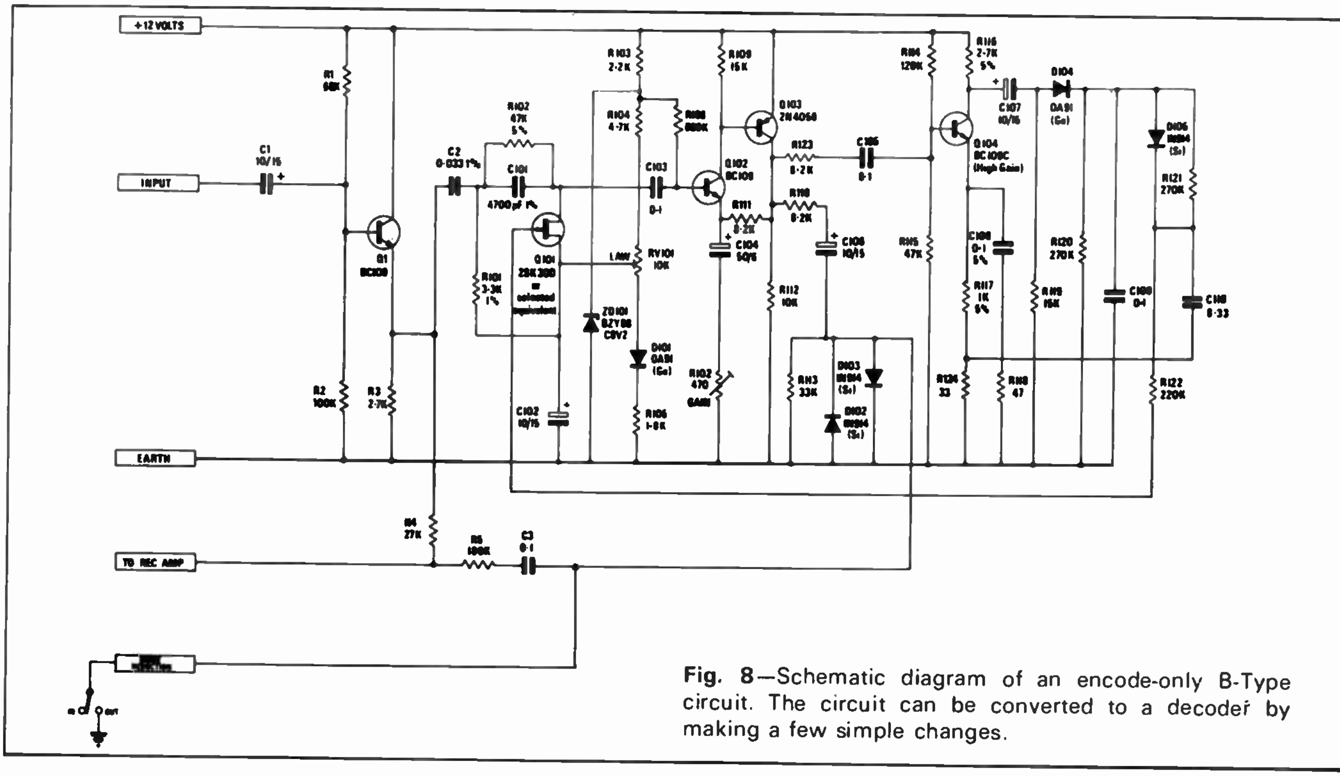

Fig. 8—Schematic diagram of an encode-only B-Type circuit. The circuit can be converted to a decoder by making a few simple changes.

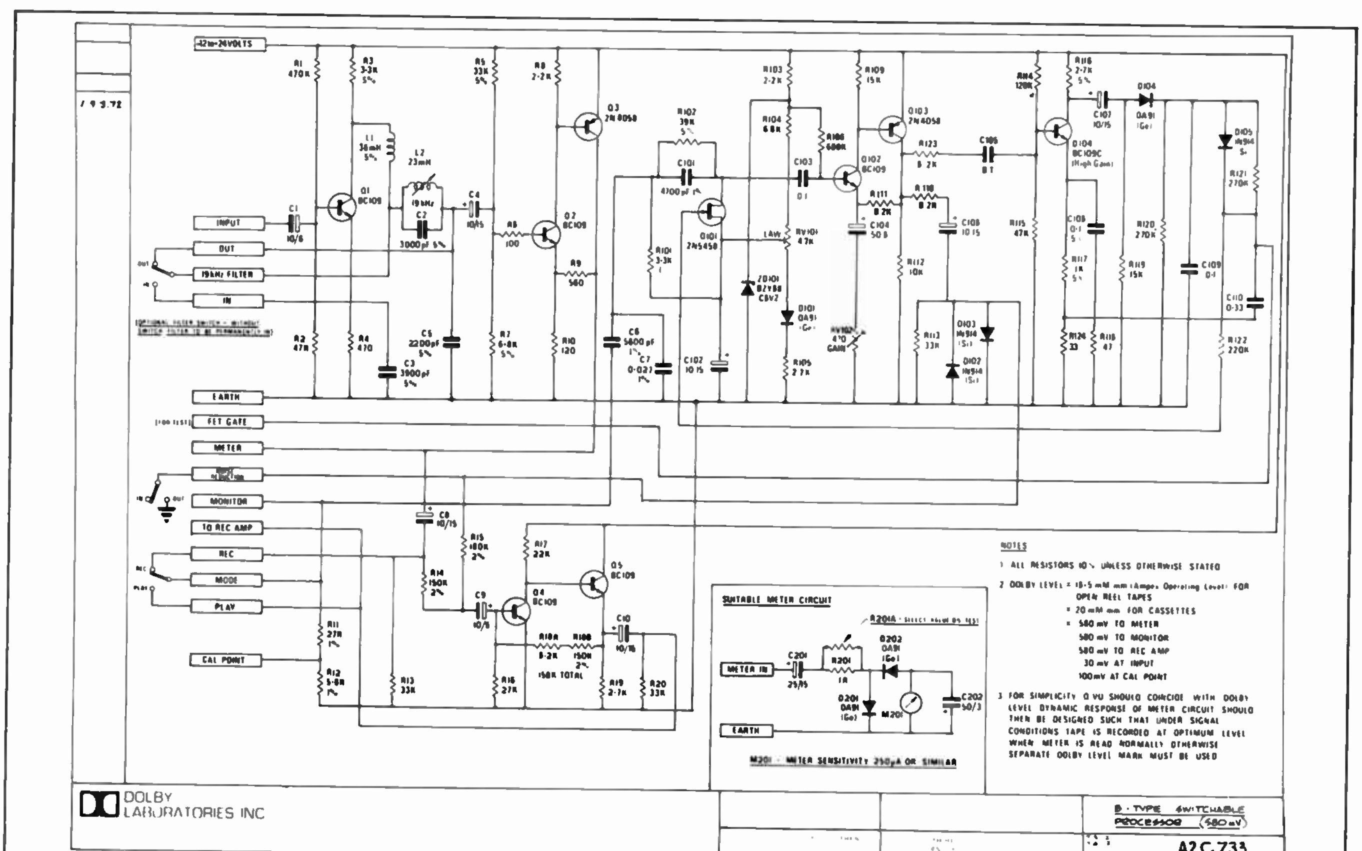

Fig. 9—Schematic diagram of complete switchable encode/decode circuit for use in tape recorder, including HF filtering, meter drive, monitor output and 26 dB of gain. Only a few more parts need be added to make complete record/play electronics for one channel of the recorder.

diodes in the overshoot suppressor stage limit the peaks of the overshoot. Since this takes place in the secondary path, the result of the suppressor action is to limit overshoot to a relatively small fraction of the full-level main path signal. Further, by restricting overshoot suppression to the secondary path, it is possible to avoid introducing audible distortion to the encoded signal. Because a complementary action takes place during decoding, the small remaining overshoot in the encoded signal is eliminated, and as with other effects produced during encoding, the original signal is restored. Figure 7 shows the result of encoding and decoding a short burst of 3 kHz, which changes in level from −40 dB to +6 dB.

Figure 8 is a typical schematic diagram of an encode-only B-Type circuit; the circuit for decoding-only is similar. As can be seen, only five transistors plus an FET are required; the parts cost of the circuit is approximately $2.40.

Figure 9 is the schematic diagram of a B-Type processor which has been designed to integrate noise reduction with other tape recorder electronics requirements as much as possible. The resulting circuit provides 26 dB of gain, whether or not noise reduction is in use, bias and multiplex filtering, and meter and monitor amplifiers. In fact, the only additional electronics needed to complete the recorder are a bias oscillator, recording amplifier (one transistor) and a microphone and head amplifier (two transistors). With the active elements used in the record/play switchable processor shown (eight transistors and one FET), the total used in the recorder, for two channels, is 22 transistors and two FET's. The cost to a manufacturer of the components shown in Fig. 9 is about $3.20, excluding the bias and multiplex filter components, which are, of course, necessary in the circuits of any properly designed tuner and recorder.

Dolby Laboratories and Signetics have collaborated in the development of an integrated-circuit version of the B-Type circuit. The IC is expected to offer manufacturers economy of assembly, elimination of adjustments, and somewhat smaller space requirement than the discrete-component version.

The characteristics of Dolby B-Type noise reduction can be summarized as follows:

1. Program recovery characteristics, with regard to frequency response, phase response, transients, and signal dynamics, are theoretically perfect; in practice, this ideal is attainable to any desired accuracy. Distortion in practical B-Type circuitry is considerably lower than that of the tape recorders or tuners with which it is used. Any type of program material can be encoded and decoded without audible loss.

2. The circuit is simple, inexpensive, and small in size, either in discrete-component or IC form.

3. The circuit is easy to manufacture and use because of the absence of critical components or adjustments. The circuit can be quickly and easily calibrated during manufacture, after which further calibration is not required. In use, only a simple level adjustment is necessary if tape of significantly different sensitivity is substituted for that formerly used.

4. No modification of broadcasting or duplicating practice is required to incorporate B-Type encoding. The use of the noise reduction system often makes worthwhile other improvements, however, such as extension of frequency response and dynamic range, or reduction of distortion by use of lower modulation levels, or some combination of these.

Effects Upon Noise Spectra

Figure 10 is a multiple exposure of the screen of a 1/3-octave real-time analyzer, allowing a direct comparison of the noise spectra at the output of a high-quality cassette recorder when different kinds of tape were used with and without the Dolby B-Type noise reduction circuit. Curve 1 is that produced by C90 ferric oxide tape; curve 2 is that of C90 chromium dioxide tape; curve 3 is produced by the same tape used for curve 1, but the B-Type circuit is switched "in," and curve 4 represents the noise spectrum of the chromium dioxide tape with the circuit in. The tapes shown were biased before the measurements were made; no changes in gain or other control settings were made during the tests, other than to set equalization differently for the chromium dioxide tape from (70 microsecond). In fact, most of the improvement in noise level obtained when chromium dioxide tape is used appears to be due to the change in equalization; if this change is not made, there is little advantage in chromium dioxide tape from a noise point of view. On the other hand, the combination of chromium dioxide tape, 70 microsecond equalization, and B-Type noise reduction results in an excellent noise figure, 57 dB below Dolby Level in the example in the photograph (DIN 45405).

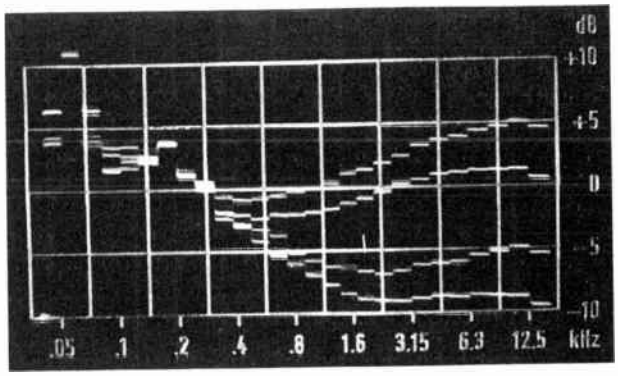

Fig. 10—Noise spectra at output of cassette recorder during playback of four kinds of tape. All tape was biased and control settings were the same for the four tests. (1) Ferric oxide C90 low-noise, −45.7 dB DIN 45405; (2) Chromium dioxide C90, −48 dB; (3) Same as (1) with B-Type noise reduction, −55 dB; (4) Same as (2) with B-Type noise reduction, −57.1 dB. All noise measurements referred to Dolby Level = 200 nWb/m.

The advantages of B-Type noise reduction are also obtained when the system is used for FM broadcast transmission and reception, i.e., the improvement in signal-to-noise ratio obtained by use of the B-Type circuit is approximately the same as that produced by a 10 dB increase in field strength. The significance of this improvement can be appreciated when it is realized that such an increase would usually require an increase in transmitter power by a factor of ten. Considerable experimentation and broadcast experience in the USA have demonstrated, as one would expect, that the area in which listening is satisfactory is greatly extended by use of the B-Type noise reduction system. Several American classical music FM stations are already broadcasting full-time using Dolby B-encoding.

Compatibility

When any improvement is made in a system as widely used as the compact cassette system, it is highly important that the new development should be fully compatible with existing equipment. Improved cassettes must be playable on any machine which can play old-type cassettes, and fortunately this is true of Dolby B-Type cassettes. Such cassettes are subjectively compatible (i.e., generally pleasing to the listener) when played without decoding circuitry, to a great extent because of the unique approach taken in the B-Type circuit.

Because most low-cost cassette machines are deficient in high-frequency response, the increase in low-level high frequency content in a B-Type cassette is usually welcomed by listeners with such equipment. Cassette recorders of higher quality, or the associated equipment with which they are used, contain tone controls which permit the balance to be adjusted to suit the taste of the listener. It is quite likely that many of the millions of B-Type encoded cassettes which have been made commercially are owned and played by listeners who are unaware of the special nature of the program material they hear. In any case, the subjective difference between encoded and other cassettes is sufficiently unobtrusive that none of the recording companies offering "Dolbyized" cassettes have found it necessary to offer old-type cassettes as an alternative.

It is worth noting that almost all pre-recorded cassettes are already compressed, for only in this way can the audibility of low-level passages be preserved in programs of wide dynamic range. B-Type cassettes differ mainly in that the listener now is able to remove the compression by pushing a button on his cassette machine restoring program dynamics and reducing noise. This is only possible because B-Type compression is standardized, while other types of compression vary considerably.

Commercial Use

Within a few years of its introduction, the Dolby B-Type noise reduction system has been licensed to most major manufacturers of consumer tape recorders. At the present time there are more than 40 licensees manufacturing over 100 different B-Type products. Licensee payments for use of the circuit are on a sliding scale, based on quantity, from a maximum of 50¢ (U.S.) to 10¢ per circuit. Royalty charges are typically 60¢ per stereo unit for a major manufacturer.

In addition, most of the pre-recorded cassettes now made in the United States, the United Kingdom and Japan are "Dolbyized," and many of the largest recording companies issue their cassette output in this form, among them Ampex and CBS in the United States, Decca and RCA in England, and CBS-Sony, Nippon Columbia, King, and Apollon in Japan. Pre-recorded open-reel tapes and 8-track cartridges are also becoming available. In the United States, a number of FM stations have already started to broadcast regularly in B-Type encoded form, and this procedure is under study in other countries as well. There is no royalty payable for encoding cassettes or other tape recordings, or broadcasts.

Conclusions

The reduction of background noise by the Dolby B-Type noise reduction system has contributed importantly to the improvement in quality of home tape recording and playback. It has helped to make the extension of frequency response, the reduction of wow and flutter, and other improvements worthwhile, particularly in cassettes. The unique characteristics of the B-Type system permit excellent noise reduction without program losses, noise modulation and other drawbacks which have afflicted earlier attempts to solve the noise problem. The simplicity and economy of the B-Type circuit facilitate its use in consumer products at all price levels. Æ

Noise Reduction Techniques

H. W. Hellyer*

LET'S TAKE A LOOK at one or two ventures into noise exclusion that have been at least a bit more ambitious than a mere clipping of playback peaks. One such system is Panasonic's NFD device. NFD, quite simply, mutes the line output unless the signals (on playback) are above a predetermined level and below a set frequency. This reduces hiss when the signal level is low. That is, you get what you want when you most want it.

In the RS 735US, there was a two-transistor, nine-diode circuit that gave very good results indeed. Figure 1 shows the basic configuration. Signal-to-noise ratio, when I tested it, with this noise filter employed, was as good as 66.5 dB. At 1 kHz, the improvement was a mere ¾ dB, but although at rated output level the NFD only made 1 dB difference to the S/N ratio, when the level of signal was down around the danger level, approaching what would have been obtrusive hiss, the circuit effectively blanked signal, and its action did not, as with so many compandor systems, provide an aural switchback.

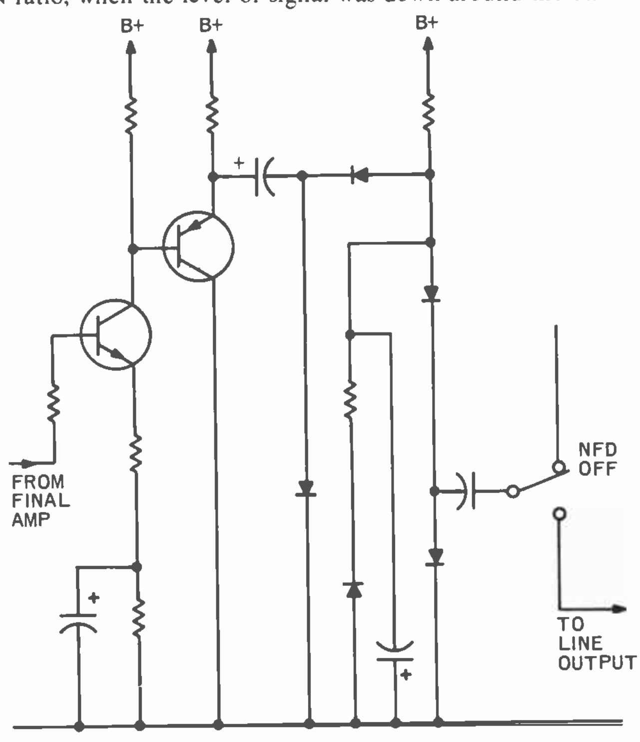

Fig. 1—A simple muting circuit used by Panasonic—simple, but effective, sensing the signal level and "killing" the line output when the signal drops dangerously near the noise level. The circuit shown is for one channel. The same two-transistor network is employed for the other channel, and this "commoning" can lead to problems.

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Taking the replay system a step farther, Philips has the DNL innovation, which should make much cassette work with other folk's tapes a really feasible possibility.

DNL means Dynamic Noise Limiter, and Philips (Norelco to you) argues thus . . .

"When music is played softly, it is made up almost entirely of pure tones in the middle and low frequency ranges with hardly any harmonics. This is mainly because very few musical instruments produce tones whose fundamental frequencies are much higher than 4.5 kHz. Tape hiss, however, is made up of sounds in the higher frequencies so that it is during the soft passages and silent intervals that it becomes most noticeable.

"When music is played loudly, it not only contains the lower and middle frequency pure tones, but also a great deal of harmonics, which give character to the sound. It is in the loud passages that noise suppression is unnecessary as the high frequency harmonics hide the tape hiss. Any filter action would make the music sound dull and unnatural.

"Therefore, if tape noise or hiss is to be suppressed, it must be completely eliminated in periods of no music signal, reduced during the soft passages of music, and left unsuppressed during the loud passages."

Thus, the oracle—begging one or two questions, like: "Pure tones—all instruments played softly?" and "What happens to the soft tones of one instrument when another plays loudly?" and "How soon after the loud noise ends does the suppression take place?"

The Dynamic Noise Limiter acts on replay, the argument being that it therefore allows complete compatibility, giving the benefit of noise suppression even to those poor, deprived owners of untailored cassettes. It is, effectively, a steep, low-pass filter which acts when there are no high signal frequencies.

Philips has been rather clever about it, allowing high frequency signals that exceed a predetermined level to bypass the filter: so there are two signal chains. Fig. 2 shows the block diagram. From the splitter, the signal takes two paths, one path merely inverts the phase without affecting the linearity while the other passes it through the tailoring process.

This process chops off the lower and middle frequencies, leaving only those above 4 kHz (approximately—you can't do these chopping actions abruptly without introducing almost ineradicable distortions, whatever the advertising copywriters say). This remaining high frequency band is now monitored so that the quieter parts of higher frequency are boosted. Hence the variable attenuator—it is both level and frequency-conscious.

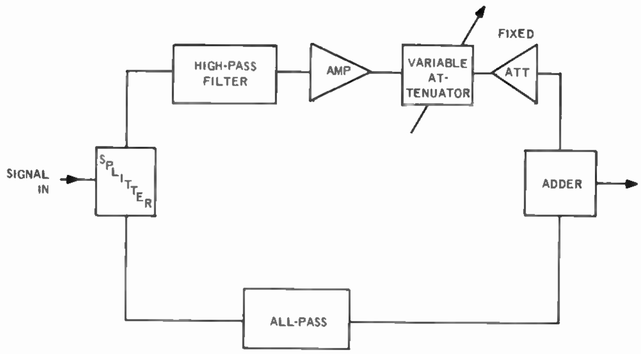


Fig. 2—Block diagram of the Philips (Norelco) Dynamic Noise Filter. The surprisingly effective though unsophisticated system acts on playback only and has the effect of an 18 dB/octave filter when the signal is low. A S/N ratio improvement of around 10 dB at 6 kHz and 20 dB at 10 kHz has been measured (unweighted). The high-pass filter takes effect above 4 kHz.

Fig. 3—The DNL circuit, four transistors, six diodes, and a handful of common components, can easily be made up into a neat set-side box—no bigger than a double pack of 20 cancer-sticks.

Adding together the processed and unprocessed chains should now, theoretically, give a signal whose low-level high frequencies have a quietened effect, while middle and low frequencies are unaltered and where the higher volume high frequencies are given their full, required weight. In theory, once again, the result should be a true replica of the original, but without the hiss.

And, I must admit, despite some initial misgivings because Philips demonstrated this device to us a year or so ago in an hotel room whose air-conditioning added some 30 dB to the ambient noise, the subjective effect is a cleaner sound, whatever the condition of the recording.

But I still feel that the answer is not to use a circuit that gives, as Philips claim, a 10 dB improvement of S/N ratio at 6 kHz and a 20 dB improvement at 10 kHz on replay, but to improve the overall record/replay process in such a way as to retain its original sound structure, not "tailor" it. Again, if you must have slow-speed, narrow-track recording, then you have to engineer out the hiss, not allow it to happen and *then* try to beat it.

So we come to Dolby and the now-famous stretching process that Dr. Ray Dolby pioneered. The original "A" process aimed at beating the "breathing" that compansion procedures forced disc users to suffer and cost more than some recording companies could afford. It begins its work during recording, splitting the audio path into a direct and a rectifier chain. But the expensive "A" system did this in four bites, carving up the frequency spectrum to give differential gain depending on signal level within the frequency bands. These are: below 80 Hz, from 80 to 3,000 Hz, above 3,000 and again above 9,000 Hz.

Both hiss and hum are present in the recording process, and while hum can be relegated to one low portion of the audio spectrum, hiss is a very different problem. It obtrudes into the very region where our ears happen to be most sensitive. It has measurable components that extend way upwards into what some engineer colleagues of mine call the "annoyance passband." Any crude way of militating against hiss will mutilate the upper frequencies which we need to preserve to get the clash and tingle of a full musical experience.

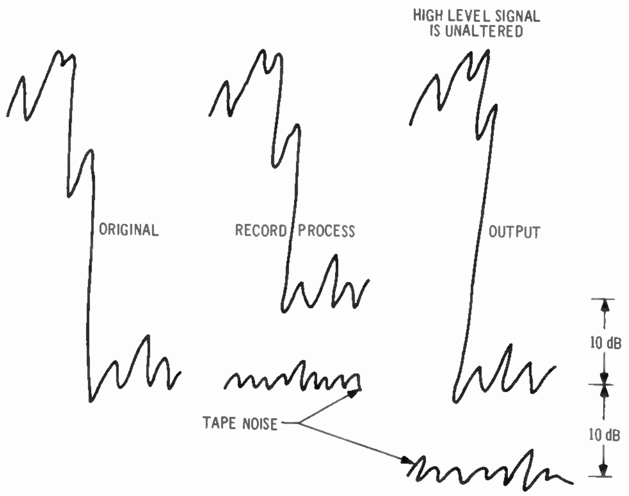


Fig. 4—One way of explaining the Dolby system: The original signal has its lower, levels down around the system noise. Processing during record gains some 10 dB of S/N ratio. Replay retains this, raising the lower levels of signal that much above the noise.

Again, the procedure is to let the noise remain when the music is loud enough to mask it. Masking—as a technical term—is a peculiar business. It depends as much on relative frequencies as on loudness, and has some strange anomalies to do with time difference and phase factors. Subject for a later discourse, maybe. At present, please take my word for it that the phenomenon happens, and by letting the main, high level signals straight through the system, Dr. Dolby follows the method we have roughly outlined already.

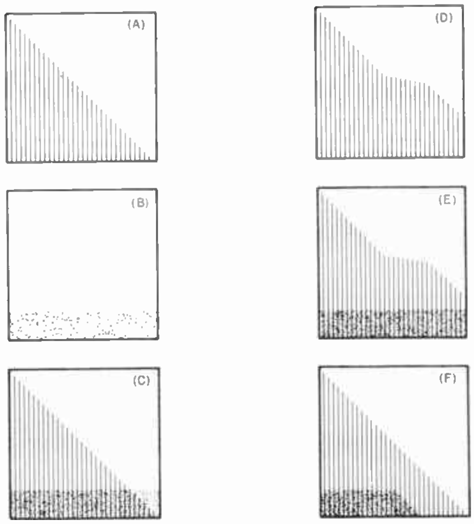


Fig. 5—An alternative explanation, as depicted by Dolby: A, music is made of sounds of different loudness with intervals of silence; B, noise of some kind is inescapable; C, when a tape recording is made and replayed the noise interferes with the low level signals, spoiling the program; D, the Dolby system boosts the lower signals during recording; E, those lower signals are still above the annoying noise during replay, as shown in F, the composite picture of the reconstituted sound with noise "reduced" by the carefully engineered boost and stretch system.

Fig. 6—Noise reduction units can be added quite easily to existing equipment. This Advent 100A has been enthusiastically received, despite the $250.00 price tag. My own special interest is harmonic distortion, and I was interested to note that the 100A was under 0.4% to 0 dB and less than 0.2% at lower levels. Output noise, −60 dB; noise reduction around 10 dB above 4 kHz, about 3 dB at 6 kHz. This is a stereo unit and well worth considering for slow-speed recording.

The subtlety lies in the treatment of the low-level signals, where noise is obtrusive. Dolby calls this the differential component, and this is, of course, relatively small—and hence more difficult to handle. It has to be remembered that the noise reduction system does not eradicate noise; it boosts weak signals to improve the signal-to-noise ratio, that's all.

That's all! Pause for hollow laughter! Arguable decisions are the threshold limit, below which noise-plus-signal will be processed, attack time, the response of filter circuitry to the information that a signal in need of treatment is coming along, the amount and nature of compression, and the way of ensuring a mirror image expansion and an avoidance of overshoot (which would process signals that did not need such treatment).

If the distortion has a duration of less than a millisecond, it will defeat the human ear. This is a smaller fraction than normal signal transients and our aural loudness-growth characteristic cannot distinguish the short-lived distortion.

The Dolby "B" system came into being when Ray Dolby was asked to dream up a modified noise reduction device for use with domestic equipment. The only feasible way to keep such a system within our budget was to forgo the technical requirement of four passbands and operate over the whole audio spectrum, this time making the sensor part of the apparatus listen for frequency as well as loudness, on a kind of sliding scale.

The system comes into action at about 600 Hz, with a maximum 3 dB effect. (O.K., so the ads say it extends above 2 kHz, but the sliding scale method means it really begins lower down). At 1.2 kHz it has a maximum 6 dB effect, has 9 dB at 2.4 kHz and reaches the advertised 10 dB above 4 kHz. The compression comes in about 45 dB below what has become known as the "Dolby level." This can be defined as a flux level on tape of 200 nanowebers per meter. Call this 0 VU.

In more technical terms, the differential chain splits into the rectifier path and into the linear path to the mixer for readdition to the main signal. The rectifier path contains boost circuits giving a 6 dB per octave flip to the higher frequencies. Then the output is rectified. This rectified signal effectively alters the dynamic resistance of an FET at the input end of the chain, and so gives a boost at low dynamic levels and practically no boost at high levels. By the simple device of driving the FET via a small coupling capacitor, Dr. Dolby achieved both a drop in gain with an increase in dynamic level and a change of the turnover frequency of the "threshold" as the level changes. The sliding scale, in fact.

At low levels the capacitor lets the FET see the full signal and gives a good 10 dB boost above 2 kHz. Increase the input level and the frequency above which this full boost is given begins to rise. Turn up the wick still more and the treble boost in the rectifier chain stops the over-saturation of the tape. To reinterpret, that means the tape is driven to its full limit when need be, at high dynamic levels (of original signal), but is allowed up to a 10 dB boost at lower signal levels. The replay mode is reciprocal.

The entire processed chain is inserted in a feedback loop around the main chain to subtract instead of add. The elegance of the system is that the same basic circuitry, and, indeed, a mirror-image printed circuit board makes production costs tumble and the add-on Dolby units now available should be within any enthusiast's purse. (Dolby IC chips are also coming soon.—Ed.)

Fig. 7—Slim, elegant, technically precise, one section of the Dolby A system as used by professional recording bodies throughout the world. Having had the chance to "rip one to bits," I can vouch for its engineering excellence.

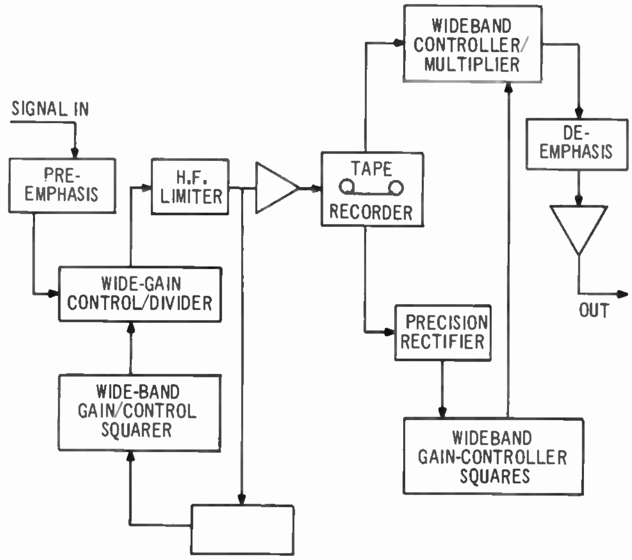


Fig. 8—Block diagram of the Burwen system, with refinements like active transformers and direct play equalizers omitted. The heart of the system is the rectifier module, monitoring the gain of two channels simultaneously in the "domestic" system. Operational amplifiers are used widely in this system with very high accuracy as a result.

My own tests with those available in the U.K. have confirmed that signal-processing of cassette-recorded music, speech, and sound effects have done wonders to guard against hiss and have not made detectable any audible worsening of the prime signal.

After Dolby, what? Well, according to Richard Burwen, quite a lot. In the December, 1971 issue of the Journal of the Audio Engineering Society, I came across the Design of a Noise Eliminator System which gave me much brain-searching and is at present exercising the pundits in those polite tomahawkeries of the erudite correspondence columns. (See also AUDIO, June, 1971.)

To begin with, the title of Richard S. Burwen's paper hits a sore point. The only way you eliminate noise, truly, is not to cause it. After the die is cast, all you can do is guard against it—which we have seen three different systems doing in the preceding notes.

Mr. Burwen took the critics by the ears at the 41st convention in New York on October 5, 1971. In February of that year, a paper of his entitled "A Dynamic Noise Filter" had aroused comment. He is more concerned with studio tape machines, just as Dr. Dolby was, and there seems little hope, at present, of such an elegant "domestic" solution to the noise reduction problem with a plain man's Burwen. But anyone who has been in the audio field as long as us (well, me) knows better than to say that something, anything, cannot be done.

So let's conclude with a brief look at Mr. Burwen's solution.

He set himself some pretty high parameters. His system was not, he told us, to exceed the present 1%, and preferably 0.5%, distortion level of good taping. He wants to record live music "with no audible noise whatsoever." So his first experiments were to determine peak recording levels.

Recording to +3 VU, a normal process, when 0 VU is the standard set limit and peaks above this as much as +6 VU are occasionally tolerated because of their short duration, meant that distortion on tape went over that critical 1%. He concluded—first point, and first stumbling block for his critics—that it is not always advisable to retain every peak.

Listening tests revealed that for noise to be negligible in the absence of program material, it had to be 90 dB or more below the 1% distortion level, i.e., better than -84 VU. Then he found that noise 65 dB down was audible with a 500 Hz sine wave but masked by frequencies above 3 kHz. You could reduce the bandwidth to about a half-octave centered on 500 Hz and get a pure tone—so the solution seemed to be split the waveband, per Dolby A.

But the multiband system, according to Richard Burwen, has the disadvantage of frequency response errors in the tape machine causing errors in the expansion process. The solution was to use the whole band but compress the 90 dB expected input to 30 dB at the tape. He then combined the principles of his dynamic noise filter (see June, 1972 issue of this magazine) with a single wideband compandor.

The dynamic noise filter acts as a low level expander at top and bottom of the frequency spectrum—again, something like we've seen before. Adding a high and low-frequency compression system seemed to be the answer, and high frequency pre-emphasis was intended to improve the S/N ratio. Some hellish problems raised themselves at this point, and Mr. Burwen went back to the drawing board. He finally produced three systems, A, B and C. Characteristic A is optimized for studio recording at 15 ips. It has a dynamic range of 110 dB and this is the one you'll see hailed in the ads! System B operates more modestly to give a 102 dB dynamic range at 7½ ips, and C is the one that may eventually interest us at 3¾ or 1⅞ ips for FM broadcasting, records or background music. If you want it in the words of the master: "The system . . . utilizes high and low frequency pre-emphasis and a single wideband cube root compressor to produce the recorded signal, and a complementary expandor and pre-emphasis for playback."

The important point slipped in later is that in the single-band system the frequency response is constant and is not affected by inaccuracy in the tape machine. Again, we shall leave the pundits to argue.

The high performance of the Burwen circuitry has been made possible by the low-noise two-quadrant multiplier/divider. Bettering Dolby by one magnitude in claim and applicable also to FM systems, it seems to offer possibilities, and we must wait and see what the outcome may be.

For my part, in this noise-polluted environment, I welcome any device that can help rid us of clamour. But noise is what you make it, and the tick of an obtrusive clock, as many an amateur recordist has found, can be as bothersome as a traction engine. The subjective results, applied to cassette, have been enormous—praise to the noise-breakers! Æ

TAPE RECORDER MAINTENANCE Regulators

H. W. Hellyer

AT THE END of my last contribution (Jan., 1971) I gave a hint of things to come, while referring to methods of motor control. This applied primarily to portable cassette tape recorders, and one small circuit was given, that used by Philips in their EL3302A and similar models. Researching the subject for this continuation, and also for a lecture project I had to do at the request of my old friend Donald Aldous, (see London Letter, Page 10, AUDIO, Sept., 1970)

I became convinced that the subject was worthy of a much deeper treatment than was possible as one article in this series. Our Editor (whom St. Cecilia preserve) is in apparent agreement, so I propose to go back to the beginnings and discuss the principles of regulation before leading on to some of the practices.

Reasons for this digression? They are twofold. In the first place, at the aforesaid lecture, when I took along a whole bunch of portable machines to describe and discuss with the lively South Devon Tape Recording Club, of which Aldous is the very active president, I was inundated with questions about the more sophisticated methods of servo control and voltage regulation that some of these tape recorders employ. In the second place, it was evident from some of the questions there and in my correspondence with readers of a number of audio magazines in this country (Great Britain) that understanding of the circuitry is incomplete. Power supplies are too often taken for granted. Either it goes or it doesn't.

Trouble is when it doesn't go, finding the reason is not always easy. It helps to have a knowledge of what a circuit should do before we can ferret out the causes for the stoppage.

Regulation—what do we mean? I would define this as "keeping the supply voltage constant even though the current drawn by the load varies." You may wish to apply a more elegant definition, or tie me down scientifically, but basically, this is what regulator circuits of tape recorders are designed to do. The bite comes with that last word "varies." The ordinary tape recorder has three functions: RECORD, PLAY, and FAST WIND. The first two may require much the same current, but usually the fast winding process demands more power, and the strain on the supply is more evident. Some machines need varying amounts of current for forward and reverse, even for differences between the amount of tape spooled. Simple regulation is not going to help them much when demands vary widely.

Alternative methods have motors which regulate themselves, and the supply line to the main machine can be separately and simply regulated. There are some interesting circuits in this group and we shall take a look at them later. Others employ what is now known as "servo control," where a sensing circuit picks up a mechanically generated pulse as the motor rotates, compares this with an electronically generated reference, and from the error reading feeds back to the motor a controlling change in supply voltage. Again, we have some intriguing variations.

The simplest form of regulation is the large capacitor across the battery supply. The function of this fellow is simply to bypass audio signals. It is a device to be found quite often in cheap radios as well as the more sophisticated gear we are talking about, and it does precisely the same job. In more technical terms, it reduces the apparent source impedance.

The source impedance of an ordinary layer-type battery could be a couple of ohms, and this will get bigger as the battery ages. At 100 Hz, a 1,000 μF capacitor will have an impedance only

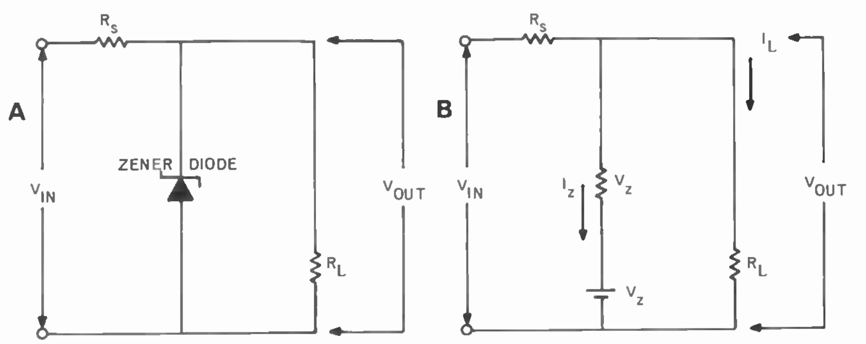


Fig. 1—Zener diode regulation, showing **A** the actual and **B** the effective circuit of the zener-controlled power supply.

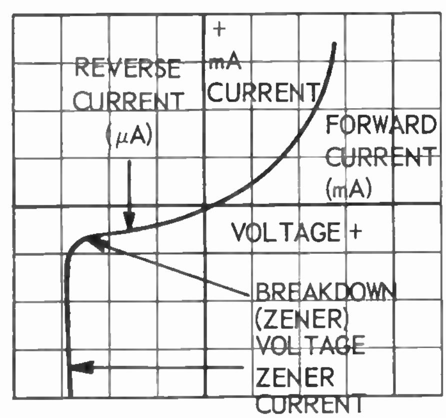


Fig. 2—Characteristic curve of a typical zener diode. Values are not marked on the axes of the graph, as these are different for various grades and specific values would be misleading. Note that reverse current is in microamps.

controlling transistor, as we saw from the circuit of the Philips design, at the end of the last article. By careful selection of the source voltage, i.e. its circuit derivation, the inherent flywheel action of regulation can be used to advantage. Figure 6 gives a British Radio Corp. modification to the Philips design shown in the Jan., 1971 issue of AUDIO. Note that here we have the control voltage for the base of the first transistor derived from a potentiometer network, with a preset variable, and the upper end of this chain is taken to the collector of Tr2. The circuit has two functions: voltage stabilization and motor regulation.

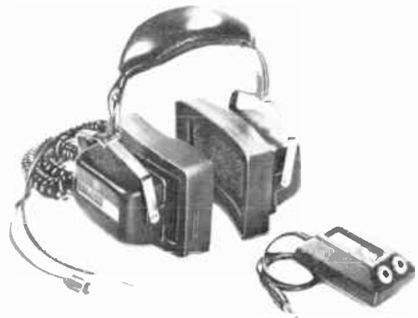
With everything normal, Tr2 base bias is determined by Tr1 collector current. This is set so that the correct voltage is fed to the motor while the effective forward resistance is lower than the 470 ohms in the emitter of Tr1. The two diodes, used here as temperature compensated devices and mounted near the heat-sink of Tr2, are forward biased to the constant voltage portion of their characteristics. A drop in supply voltage affects the emitter of Tr1 via the forward biased diodes. A smaller effect is felt at the base by the potentiometer chain so there is a net increase in forward bias. The collector current of Tr1 rises, driving Tr2 on harder and reducing its effective series resistance, which offsets the voltage drop in the supply.

As a motor governor, the circuit works the other way around, offsetting the change in potential across the motor which will be caused when a varying load affects the armature current. As the motor current rises, there will be an increased voltage drop across the paralleled pair of resistors (3 and 22 ohms). (Note that the 680 ohms resistor is here to give Tr1 its forward bias when the circuit is first switched on; without this, both transistors would remain cut-off.) Applied to the effective diode (base-emitter) of Tr1 via the two paths previously mentioned, it causes an increase again of forward bias, increases collector current, turns Tr2 on harder and feeds more voltage to the laboring motor. Variations of Tr2 base voltage around a mean value are ensured by the CR combination, the large value of the electrolytic helping to maintain the necessary average bias level.

Many more interesting circuits can be found, and in the next part we shall look at motor control and servo circuits in greater detail, using some of the information already gained and introducing one or two fresh concepts. Æ

Headphones

AKG K-180



David Clark 4CH-A

Fisher QP-44

MANUFACTURER	MODEL	Type	Frequency response, Hz ± ? dB	Impedance, ohms	Sensitivity mW input for 100 dBm out	Maximum input, mW	Distortion, %	Cord length, ft.	Weight, oz.	Price	SPECIAL FEATURES
AKG (PHILIPS)	K-100	Dyn	20-20k ± 3	600	1		1.0	8	13	29 00	
	K 150	Dyn	20-20k ± 2	600	1		1.0	6	9	39 00	
	K-180	Dyn	20-20k ± 2	600	1		1 0	8	21	69.00	
	K-158	Dyn	20-20k ± 2	600	1		1.0	6	15	60.00	Boom set; 2 chan., plus bal . low imp. mic
AVID	Pro 730V	Dyn	20 18.5k	300	0 3*	330 (10V)	0 5	15	16	29.95	Vol contl for ea chan. on headset; mylar diaphragm; *for 100 dB
	PRO 730	Dyn.	20 18 5k	300	0 3*	330 (10V)	0 5	15	16	26.95	Mylar diaphragm. *for 100 dB.
	885	Dyn.	30-17.5k	300	0.3*	330 (10V)	1 0	15	15	14.95	Same as above; *for 100 dB
BEYER DYNAMIC (REVOX)	DT 100	Dyn.	30-20k	2x200	*	1000	0 2	10	9	57.50	*100 dB/mW over 2x10⁴ µbar at 400 Hz; supplied w/ coil cbl.; avail. 2x5, 2x400, 2x1000 imp.
	DT 480	Dyn	20-20k	2x200	*	1000	0.1	10	16	80.00	Supplied w/coil cbl.: avail. 2x5, 2x25 imp.: *115dB/mW over 2x10⁴ µbar at 400 Hz.
	DT 48	Dyn	16-20k	2x200	*	6330	0.1	6½	17	110 00	*112 dB/mW over 2x10⁴ µbar at 400 Hz; avail. 2x8 or 1x25 (mono) imp.; DT 48K w/coil. cbl., $115.00.
	DT 204 4-chan.	Dyn	20-20k	4x200 (4-ch.) 2x100 (2-ch.)	*			10	14	104 00	*114dB/mW over 2x10⁴ µbar at 400 Hz.
DAVID CLARK CO.	4CH-A 4-chan.	Dyn.	40-16k	16	80	600	2.0	12	16	80.00	With DC-2A decoder. $95.00 Allos 4-ch. sound used w/2-ch. sys. and 4-ch. encoded discs.
	100A	Dyn	20 18k ± 6	17	105	1000	1 2	9	16	50 00	Also avail in 300 & 600 ohm.
	200	Dyn	30-16k	8	100	500	1.5	9	16	29.00	Clark 250 same as 200 w/vol contl.; $34.00.
	300	Dyn	40-14k	8	100	500	2 0	9	16	21.00	
	75	Perm Mag.	40-16k	8	105	1000	2.0	8	8	14.95	
FISHER	HP-70	Dyn.	30-18k	16	2.5	500		10	12	29.95	
	HP-100	Dyn.	18-22k	50	2.0	700		8	10	49.95	
	QP-44 4-chan.	Dyn.	20-18k	8	10	200		10	20	69.95	
GTE SYLVANIA	SP20BN	Dyn	20-20k	16	3	1000	1.0	14	14	19.95	
	SP40	Dyn.	20-20k	8	1	700	1.0	8	24	39 95	
JVC	STH 10E	Dyn.	20-20k	8		500	1.0	7	13	29.95	2 built-in vol. contls
	5944 4-chan	Dyn.	20-20k	8		100	0.5	7	21	49.95	Built in phase rev swit
JANSZEN	Jecklin	ES	30-18k ± 3	8	10 watts	80 watts	0.5	10	22	300.00	No air-seal to ear Add'l. set, $190.00.
KLH	80	Dyn	20-20k ± 4	600	0 06	1.66	0 5	10	11¼	49 95	For use w/amp imp of 0 to 600 ohms

Headphones

Lafayette F-4400

Pickering PH-4955

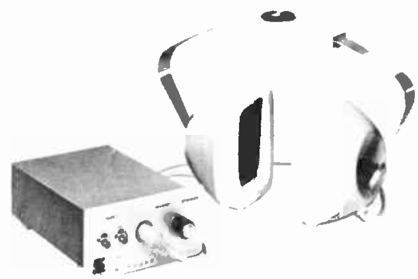
Koss Travler

Pioneer SE-405

MANUFACTURER	MODEL	Type	Frequency response, Hz ± 2 dB	Impedance, ohms	Sensitivity mW input for 100 dBm out	Maximum input, mW	Distortion, %	Cord length, ft.	Weight, oz.	Price	SPECIAL FEATURES
KOSS	ESP/9	ES	15-15k ±2	4-16			0.2		19	175.00	W. E-9 energizer. ESP/6A similar but w. T-3 self-energizer; resp. 30-19k ±5 dB; $95.00.
	Pro/4AA	Dyn.	10-20k	3.2-600				10	19	60.00	PRO/600AA similar but 600 ohms impedance, $65.00.
	KO/747	Dyn.	30-20k	3.2-600			1	10	21	45.00	KO/747Q, 4-chan.; 20-19K resp., $60.00.
	KO/727	Dyn.	10-18k	3.2-600				10	19	34.95	
	KRD/711 Red Devil	Dyn.	10-17k	3.2-600			0.5	10	12	24.95	K/711 similar but black.
	K/6LC	Dyn.	10-16k	3.2-600				10	17	29.95	Slide Vol. contrls. K/6 similar but no vol. contls., $22 50. K/6LCQ, 4-chan.; 20-17k resp.; $45.00.
	SP/3XC	Dyn.	10-14k	3.2-600				10	15	15.95	
	Travler	Dyn.	20-18k	3.2-600				8	9	29.95	Compact, fold-up design.
	HV/1	Dyn.	20-20k	3.2-600			0.5	10	9.5	39.95	Vented cup; high velocity, low mass diaphragm.
	K/2+2 4-chan.	Dyn.	10-20k	3.2-600			0.5	10	22	85.00	Vol./bal. contls.; 2/4 chan swit.
	Pro/5Q 4-chan.	Dyn.	20-20k	3.2-600			0.5	10	21	75.00	Vol./bal. contls.; 2/4 chan. swit.
LAFAYETTE	F2001	ES	5-35k	8-16				10	16	49.94	4½ x 5" cushions; adj. headband; incl. energizer.
	F4400 4-chan.	Dyn.	20-20k	4-16				9½	26	44.95	
	F1000	Dyn.	20-20k	8				6	22	39.95	2 two-way 2½" transducers; ind. left & right vol. contls.
	F-600	Dyn.	20-20k	200				5	19	29.95	
MAGNAVOX	1A9217	ES	30-18k ±5	8-30	3000		0.5	10	14	99.95	Req. 110V. a.c. src.; pwr. supply incl.; provides ext. spkr. terminals.
	1A9216	Dyn.	35-17k ±5	30	3		1.0	10	8	29.95	
	2A9186	Dyn.	30-16k ±5	8	20	1500	1.0	10	18	27.95	Coiled cable; 3-pos. treble comp. swit.
	1A9163	Dyn.	40-14k	8	20		2.0	10	12	9.95	Coiled cable.
MARANTZ	SD-1	Dyn.	20-20k	8		1000	2.0	10	16	29.95	
	SE-1S	ES	20-20k +2, −5	30	3 V.		0.15	10	48	129.95	Overload protect.; includes energizer.
MURA	SP-205	Dyn.	30-20k	8		1000	1.5	15	16	54.95	Slide tone & vol. contls.
	SP-600	Dyn.	20-20k	8		1000	1.5	10	6	26.95	Open-air type.
	QP-280 4-chan.	Dyn.	20-20k	8		2000	1.5	10	16	39.95	4-chan. stereo swit.
	QP-300 4-chan.	Dyn.	20-20k	8		2000	1.0	10	24	69.95	8 spkrs.; bass refl. type encl.; x-over ntwrks.
OLSON	PH219	ES	25-19.5k ±3	8	1.2 V.		0.5	10	14	59.98	Coil cord.
	PH220 4-chan.	Dyn.	20-18.5k ±1.8	8	2.0	600	0.32	10	12	48.49	
	PH222	Dyn.	25-18k ±1.5	8	2.5	500	0.38	10	8	24.98	
	PH213	Dyn.	20-20k ±1.5	8	2.0	650	0.3	10	10	42.98	Slide vol. & tone contls for each earpiece
PML (ERCONA)	D42 Deluxe	Dyn.	30-20k	200	0.3		2.0	6	9½	29.95	Stereo or mono; imped. for mono, 400 ohms (series conn.), 100 ohms (parallel conn.)
	RDF 224	Dyn.	20-18k	8	1.0	100	1.0	8	12	24.95	Retract. coil cable w/stand. 3-cond. 'phone plug & built-in stereo/mono swit.
PICKERING	OA1	Dyn.	30-19k	8	*	300	1.0	7	10.9	19.95	*100dB at 600 Hz; open-air type; rem. contl. unit avail. #4901, $19.95.
	4933	Dyn.	60-10k ±3	8	0.11*	500	1.0	10	21	39.95	Rem. contl. avail. as above; * for 100 dB.
	4955	Dyn.	40-11k ±3	8	0.11*	500	1.0	10	28	59.95	2-way, sep. dyn. wfr. & twtr. spkrs. w/indiv. x-over networks; remote contl. avail as above; *for 100 dB.
	OA3	Dyn.	20-20k	15	0.10	200	0.5	10	7.5	39.95	Rem. contl. avail. as above; for 100 dB.
PIONEER	SE-505	Dyn.	20-20k	8		500		16	16	59.95	2-way; level and tweeter contls.
	SE-405	Dyn.	20-20k	8	113 dB 0.3	500		16	16	44.95	Level contls.
	SE-305	Dyn.	20-20k	8		500		12	14	34.95	
	SEL-40	Dyn.	20-20k	8	96dB 0.1 V	500		10	8	39.95	Open air type; alum. voice coils.

Headphones

Scintrex Supra



Stanton Isophase

Superex PEP-77D

MANUFACTURER	MODEL	Type	Frequency response, Hz ± ? dB	Impedance, ohms	Sensitivity mW input for 100 dBm out	Maximum input, mW	Distortion, %	Cord length, ft.	Weight, oz.	Price	SPECIAL FEATURES
RCA	10R201	Dyn.	20-20k	4-32	*			10	12	29.95	*110 dB @ 1 mW at 1 kHz.
	10R200	Dyn.	50-16k	4-32	*			6½	12	19.95	*108 dB @ 1 mW at 1 kHz.
	10R199	Dyn.	20-14k	4-32	*			5	10	12.95	*110 dB @ 1 mW at 1 kHz.
RADIO SHACK	Nova-44 4-chan.	Dyn.	20-20k	4-16				15		39.95	
	Pro-1	Dyn.	10-24k	4-16				10		49.95	Liquid-filled earcushions.
	Custom Pro	Dyn.	20-20k	4-16				10		24.95	Bassport design.
	HP-100	ES	20-20k	4-16						79.95	Spkr./'phone swt. on junction box.
SANSUI	SS2	Dyn.	20-18k	8		500	1.0	6	12.6	16.95	
	SS10	Dyn.	20-20k	8		500		9.8	22	32.95	Indiv. vol. contls.; 6 ft. Y ext. cord. incl.
SCINTREX	88	Dyn.	15-20k	300	4	330	0.9	14	9	24.50	
	98	Dyn.	15-20k	300	6.5	50	1.0	14	16	34.95	
	Supra	Dyn.	20-12k ±6	200	0.5	4	1.0	10	6.5	34.95	
	HQ4	Dyn.	20-16k ±6	300	1.1	14	0.7	14	24	59.95	Stereo/quad. swit.
SENNHEISER	HD414	Dyn.	40-20k ±3	2x 2000	0.64	100	1	10	5	42.95	Open-air design.
STANTON	5700	ES	30-15k ±3	4-16	2V.	*	1.0	11	15	159.95	*Prot. crt. cuts at 110dB; oper. from low imp. output of 10W rms or higher pwr. amp.; Remote contl. avail. (#5741), $19.95.
	Dynaphase 65-4C 4-chan.		20-20k	15	0.1V	1.25V	0.5	11	19	64.95	2/4 chan.; 4 spkrs.
	Dynaphase 75	Dyn.	40-11k ±3	12	0.11V.	500	1.0	13	28	74.95	Sep. vol. & tone contls.; stereo/mono swit.; sep. woofer & twtr. in each cup w/ LC x-over; rem. unit attached.
	Dynaphase 60	Dyn.	40-11k ±3	8	0.11V.	500	1.0	10	28	59.95	Sep. woofer & twtr. in each cup; LC x-over; rem. contl. avail., $19.95.
	Dynaphase 50	Dyn.	60-10k ±3	8	0.11V.	500	1.0	10	21	49.95	Vol. contls. on cups; Model 40 same but w/o vol. contls.; rem. avail., $39.95.
SUPEREX	PEP-77D	ES	10-22k ±5	4-16	3 V.	10 V.		15		120.00	Wal. console; lev. contls. for 2 chan.; self & a.c. polarization.
	PEP-79	ES		4-16	3 V.	10 V.		15		85.00	Wood-grain vinyl console.
	Pro-BVI	Dyn.	15-22.5k	4-16	10	2000		15	18	60.00	Woofer & tweeter; x-over network; acous. susp.
	QT-4B 4 chan.	Dyn.	20-18k	4-16	20	500		15	16	65.00	Stereo/quad. swit. on left earpiece.
	QT-4 4 chan.	Dyn.	25-17k	4-16	20	500		15	16	50.00	
	ST-PRO-B	Dyn.	18-22k	4-16	10	2000		7	18	50.00	Woofer & tweeter. SST, sim., 20-20k freq. resp., 15 ft. cord, vol. & HF contls., $40.00. ST-V sim., 20-18k freq. resp., 15 ft. cord, vol. contls. $30.00.
	ST-M	Dyn.	20-20k	4-16	10	2000		7	17	29.95	Treb. contls.; woofer & tweeter. ST-S sim., 30-15k freq. resp., $25.95.
	SW-2	Dyn.	30-16k	4-16	8	2000		15	15	24.95	
	ST-F	Dyn.	25-17k	4-16	20	500		15	8	24.95	
TEAC	HP-101	Dyn.	18-20k	500mV	1.0			8		39.00	
	HP-102	Dyn.	18-20k	10000	1.0			8		39.00	
TECHNICS BY PANASONIC	EAH-80A	ES	20-20k	4-16	101dB (1V., 500Hz)	5000	0.8	13	12½	79.95	Contl. adaptor incl.
TELEX	Studio I	Dyn.	20-20k ±3	3-16	105dB SPL*	1000	1.0	15	24	69.95	*at 1 kHz, 1 mW input.
	400	Dyn.	30-20k ±3	3-16	105dB SPL*	1000	1.0	15	16	44.95	*at 1 kHz, 1 mW input.
UNICORD	UHP-1	Dyn.	40-15k	50k	100	800	1.0	8	22	36.00	Battery-oper. mon. ampl. built-in; external power jack.

After so many high-fidelity and consumer publications rated our HD 414 "open-aire" headphones tops in sound, comfort and value, why would Sennheiser introduce another model?

The reason is perfection.

Not that our new HD 424 is perfect. But our engineers—the same engineers who developed our dynamic and condenser microphones for the recording industry—have made some significant advances. Enough, we feel, to warrant a new model. Enough, that a certain kind of music lover will appreciate the added fidelity, despite the added cost.

The primary difference is response. As linear as our HD 414 is, the HD 424 boasts even greater accuracy—particularly at low bass and high treble frequencies. Due to an improved transducer assembly and redesigned earpiece geometry. Heard on the HD 424, low organ notes assume an additional fundamental richness without sacrificing the "tightness" of good transient response. While violins and other high-overtone


Sennheiser HD 414 "open-aire" headphones. Rated "best" by everyone.

instruments retain the additional "transparency" their overtones produce.

No less important, especially for long listening sessions, is comfort. Retaining the "unsealed" free-air feeling so many praised in the HD 414, the new HD 424 provides even less (!) pressure on the ear, distributing it over wider, thinner acoustically transparent cushions. For this reason—and an improved, cushioned headband—the HD 424 actually seems lighter than the 5 oz. HD 414, even though it is slightly heavier.

Now, there are two Sennheiser "open-aire" headphones for you to choose from. The HD 414, rated best for sound and comfort. And a new model offering something more. That's why.

Hear them both at your Sennheiser dealer, or write us for more information. Sennheiser Electronic Corporation, 10 West 37th Street, New York 10018.

Manufacturing Plant: Bissendorf, Hannover, West Germany

Dynamic and Electrostatic Headphones

Howard Souther*

WHILE THE SUBJECT of this discourse concerns listening through headphones, to understand the subjective significance of their performance we must employ comparisons with the perception of musical sounds in real existence. Further, we will help this effort by contrasting headphones perception to that obtained from loudspeakers. Additionally, we will elaborate on the differences between the listening modes—monaural, binaural, stereo and, yes, even four-channel "surround" sound.

The earliest reproducer was a single earpiece, which included a small electomagnet consisting of a coil of wire and an iron core with a 2- or 3-inch steel plate suspended over the end of it. The electrical counterpart of the signal passing through the coil winding induced varying amounts of magnetism in the core. This in turn vibrated the suspended steel plate, moving the air and making sound. The quality of these early phones was poor because the musical range was quite restricted. However, these phones served in communications work for many years, the present telephone receiver being a not so improved version.

Monaural Sound

The sound from early phones was characterized as *monaural*, and even when two earpieces were combined into a headset, the sound had no dimension nor depth. The best description of the effect would be that of listening to the music through a window, or, perhaps, listening with one ear at the microphone location.

Binaural Sound

When two microphones are employed, spaced and separated as are one's ears and each microphone feeds its own headphone, a tremendous effect of realism is obtained, although the sound seems to be realized "in the middle of the head" and slightly above the median line of the ears. The effect is pleasing, although reality is not duplicated, since there is no movement of the head permitted relative to the sound source.

Ordinarily, recordings are not made with this microphone spacing, for to achieve dramatic spatial effects with loudspeaker playback, a different pickup technique must be invoked.

Stereo

In stereo recordings microphones are widely spaced to favor instrument groups. Stereo listening through speakers is as different from "binaural" as binaural is from real life listening.

The finest loudspeakers reproduce about 8½ octaves of the musical range out of a possible 10. When one speaker is placed to the right and the other to the left, the directional effects of the concert field are simulated in the living room. Even the sound emanating from the center of the field is synthesized by the blending of the sound from two speakers.

A cross-referencing center in the brain of the listener evaluates the intensity, quality, and arrival time of the sound from two sources in an instantaneous, composite and unconscious function. This allows him to determine that the sounds come from the left, the center, or the right. According to the loudness and reverberant quality, the listener also perceives whether the sounds are from the near distance or from far away. These things give three dimensions to the music and place the listener in the same position as the audience to produce a powerful effect of realism.

The art and science of acoustics provides tools and techniques which can enhance reality. It was Leopold Stowkowski who once said, "If I had a thousand bass viols, I could use them all." This points up one of the many limitations of real existence—and now we suggest something better.

The exclusion of distracting sounds, coupled with the proximity of the musical information received is only a part of the improvement over speakers and, yes, over real life listening too.

The plan (Figure 1) shows that however preferential his listening position, the conductor is too close to some instruments and too far from others, so that he must interpret, rather than hear, the actual musical balance. He compensates partially by bringing the soft playing harp in close, so that he can detect the delicate plucking. The inefficient violins are adjacent, with the concertmaster only arm's length away. Not so the bass viols. Their physical size and number demand a distant situation, in spite of their weak, but important low, grave tones.

The best efforts of the most famous maestro are vastly improved upon by the recording engineer when the phonograph record is produced. The recordist, using individual microphones placed in each orchestral section, augments and monitors the dim harp. He moderates the over-intensity of the clamorous

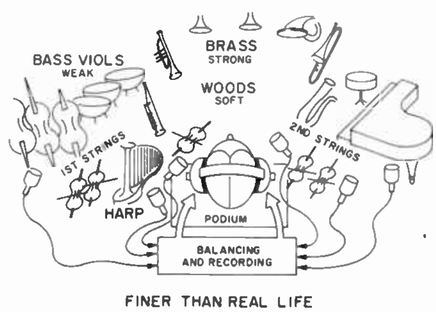


Fig. 1—This diagram illustrates the manner in which the headphone listener receives balanced sound levels from all parts of the orchestra.

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brass. He increases the normally deficient bass section, thus accomplishing a musically balanced performance almost impossible to achieve in real life.

Through stereophones we gain the finest listening position, that of the conductor himself, and through disc recordings bypass his problem of interpreting proper balance of the instruments over near and far distances.

Here, then, through stereophones, and the scientific wizardry of the acoustics engineer, is the means of appreciating music far beyond the composer's most impossible desire.

Four-Channel Sound

Currently there is extreme interest, not to mention controversy, in *four-channel sound*, where left, right, left rear, and right rear signals are recorded and played back from the four corners of the room. Peculiarly, if the listener faces the center front and holds his head very still, there is no certainty that sound comes from the rear, although the ambience effect has been enhanced, perhaps because the two extra speakers have assisted high frequency dispersion. Movement of the head, of course, permits rear directional effects. The illustration may disclose why increased "surround sound" effects are achieved with speakers, and suggest that maybe headphone stereo listening has had the "surround" effect right along.

Observe in the illustration (Figure 2) that if the listener is blindfolded and faces the center between two speakers while the head is held steady, the virtual sound originating 20 degrees to the right cannot be differentiated from sound at 167 degrees, or from *behind the listener!* (1)

If, while listening to four channels with four speakers the head is moved 30 to 40 degrees to the right, the right rear channel becomes the "right" channel, and the left rear channel the right rear channel. Thus, a new localization of the sound to the front is achieved, but the origin of the *rear* sounds remain in question. It may be interpreted that in real life a visual clue is required to determine front and rear sounds unless the head is rotated. It has also been shown that visual clues are required for determining "up" and "down" sounds. While the two ears do a creditable job of establishing left, center, and right, and the brain evaluates intensity and reverberation to determine near and far distances, to hear rear sounds (without head movement), and to hear sound from "up" or "down," we may need a total of six ears situated on each plane of the head!

Because the headphones turn with the head, movement does not assist in determining "rear" sounds. However, the superior ambience effects, which lend so much to pleasing listening, have always been present in headphones with two channel recordings and now show possibilities of enhancement with the ambience added by four channel techniques.

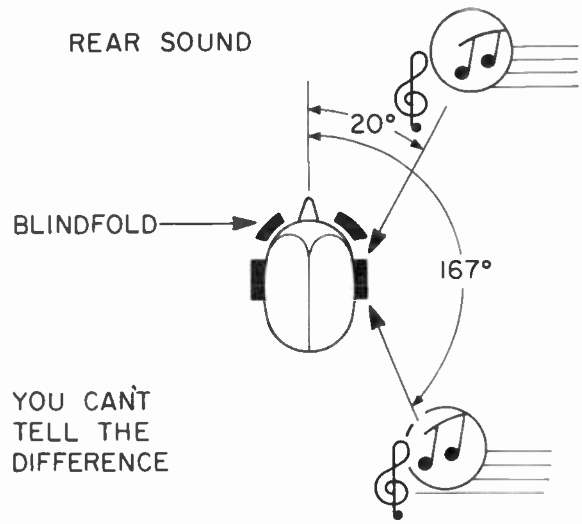


Fig. 2—When a blindfolded listener holds his head steady, rear sounds cannot be differentiated from front sounds. In this case a source 20 degrees off the axis also may be localized at 167 degrees.

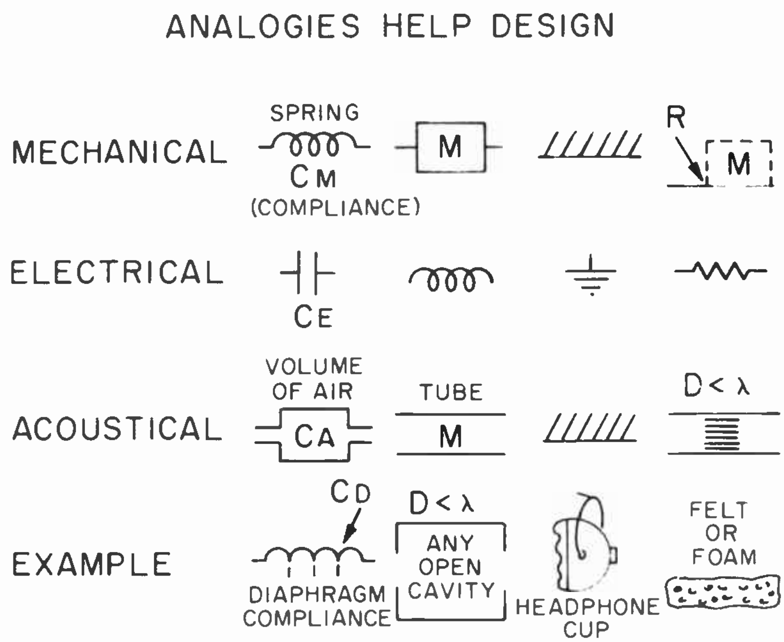


Fig. 3—Mechanical, electrical, and acoustical analogies. *Mass* is the mechanical element which opposes a change in velocity. *Inductance* is the element which opposes a change in current. *Resistance* is the electrical dissipative element. *Friction* is the mechanical dissipative element. *Capacitance* is the electrical element which opposes a change in the applied force. *Compliance* is the mechanical element which opposes a change in the applied force. It should be noted that in a mechanical or acoustical system, the capacitive element always must have one side grounded.

Tools for Design

The design of headphones is as much an art as it is a science. However, the art is ably assisted by the body of knowledge in mechanics and electricity. If this exposition serves as intended, the reader can become an expert and himself execute a good headphone design. Using as a tool the table of analogies illustrated in Figure 3 and by employing the considerations to follow as a base, a creditable headphone will result.

Before designing a headphone we must choose the method of operation. There are two of these, velocity and pressure, and we will explain the advantages of both. (See Fig. 4.)

The velocity design is light in weight, and defined as "supra-aural," which means that it rests *on* the ear. This class of phone is smooth in response and gives good sound. It does not exclude outside sounds, and thus lacks the virtue of complete involvement with the performance because of the inclusion of ambient noise.

The *pressure* operated headphone is "circum-aural." This means that the headphone cushion surrounds and seals the aural cavity. It rests on the head around the ear and scarcely touches the sensitive "pinna" or cartilagenous parts of the external ear. The circum-aural feature excludes outside noise and is very comfortable, although it makes for a somewhat heavier headset than the velocity type. The response range of pressure phones can be very wide, and for this reason they may be preferred.

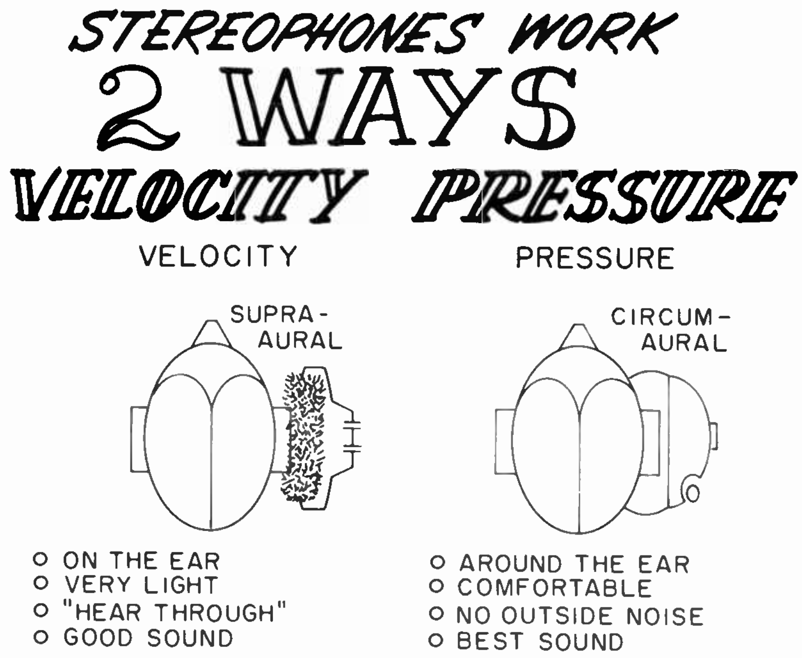


Fig. 4—Comparison of the main features of velocity and pressure type headphones.

How Velocity Phones Work

It is a nature of all physical things to vibrate at a fundamental frequency more easily than at any other. For example, a 3-inch disc of aluminum, 2 mils thick, suspended from a string in free space, if struck a sharp blow will resonate at about 1 kHz. (See Fig. 5). The mass, or inductance, and the compliance, which is analogous to electrical capacity, forms a resonant circuit resulting in high efficiency. If the disc were infinitely light and infinitely stiff (the "ideal" loudspeaker membrane operating in free space), the response would be weak at low frequencies (zero capacity or compliance) and increase at 6 dB per octave without limit as the frequency went up (zero inductance or mass). Under these circumstances, the sound would be shrill, just like that of a magnetic or velocity phonograph cartridge plugged into the auxiliary amplifier input instead of the velocity equalized cartridge input. The remedy in a velocity design is to apply mechanical equalization to our headphone to offset the shrill sound. Here is the way we go about it:

First, we select a diaphragm of the proper thickness, say 2 mils, and introduce thereby a certain amount of mass, or inductance. This mass behaves just like a coil of wire in an electrical circuit and rolls off the shrill high end. Next, we make our diaphragm compliant by having previously selected a limp material, like polyvinyl chloride, so that we add "capacity" to our equivalent electrical circuit. If we make this selection judiciously, the diaphragm will resonate where we lacked energy, or be very efficient at 200 Hz. By using foam on the frontal area of the earpiece, we introduce resistance which lowers the height of the resonant 200 Hz "hump" and controls it. This foam, or felt, distributes the pressure on the ear and gives comfort, too.

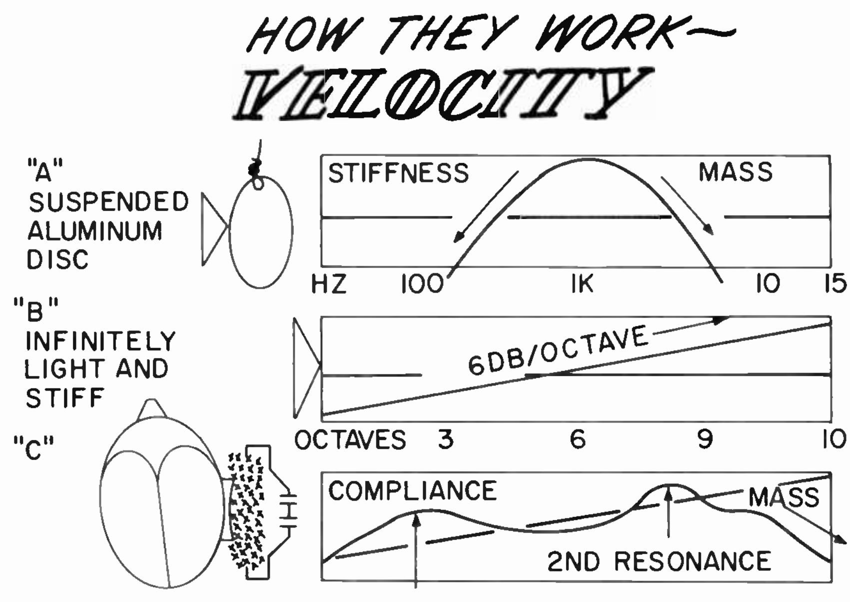


Fig. 5—A, The response of a 2-mil aluminum disc in free space when struck. **B**, The "ideal" diaphragm is infinitely light and infinitely stiff. **C**, The mass and compliance of a velocity headphone are chosen so that a large primary resonance occurs at about 200 Hz.

The diameter of the diaphragm determines the frequency of the reflection from the rim back to the center. Generally this causes a reinforcement in the treble range and is designated as the *second resonance* in the diagram. If a major reflection returns to the center out of phase, a bad cancellation of the signal energy can take place.

At this point in the design, art enters in and science leaves the scene. The physicist would say simply that you can't make a phone that will work well in these circumstances. But the engineer, by judicious employment of felt, foam, plastic, and wire, and despite humps and suckouts in the mid-range, smooths the response, struggles to get widest range and finally achieves a pleasing musical balance by minimizing the weak energy in the first three octaves and the effects of too much mass roll-off of the high end.

Pressure Type Phone

The ideal environment for propagating bass sounds is a closed cavity. Here the pressure mode of operation excells, for the circum-aural cushion seals the headphone diaphragm against the cavity of the ear. It can be predicted that response will be flat below the 400 or 500 Hz primary resonance of the system. Under these conditions we achieve linear response all the way to dc. This is a situation that can be used to help in other ways. The diaphragm can be much stiffer than in velocity phones, promoting an extension of the high end. The stiffness of 2 mil mylar, for instance, offsets the high frequency limiting mass of the material to give full energy level as high as 24 kHz.

The Glitch

We mentioned earlier that the reflection from the rim often causes a cancellation, the frequency of which depends on the diameter. This cancellation can be especially violent in a closed cavity. Referring to our chart of analogies, we find that acoustic resistance in the form of foam can eliminate peaks simply by knocking them off, or lowering the Q of the circuit. It can and does lower efficiency, too. Cone drivers common to lower cost headphones have bass efficiency to spare, but the acoustical resistance is difficult to apply, so most low cost headphones have a violent dip, or glitch, an octave or more wide in the midrange. (See Fig. 6.) This causes a displeasing hollow sound. If the glitch is more than 12 dB deep, it causes an actual physical discomfort because the brain is overstrained to supply the missing frequencies. The cone has another serious deficiency too. Not only does the mass of the cone roll off the higher frequencies, but the compliance of the very large trapped air volume shunts the high frequencies to the acoustic ground, consisting of the hard bones around the head and the headphone case.

The Fix

By using a smaller, flatter diaphragm, say two inches in diameter and mounting it close to the ear, high frequency-destroying mass and compliance (the trapped air volume in the apex of the relatively large and heavy cone) can be reduced. This reduction gives extended highs at good efficiency. The diameter glitch is still there, however, and defies almost all countermeasures.

Instead of raising the bridge we must lower the river. Removal of the glitch

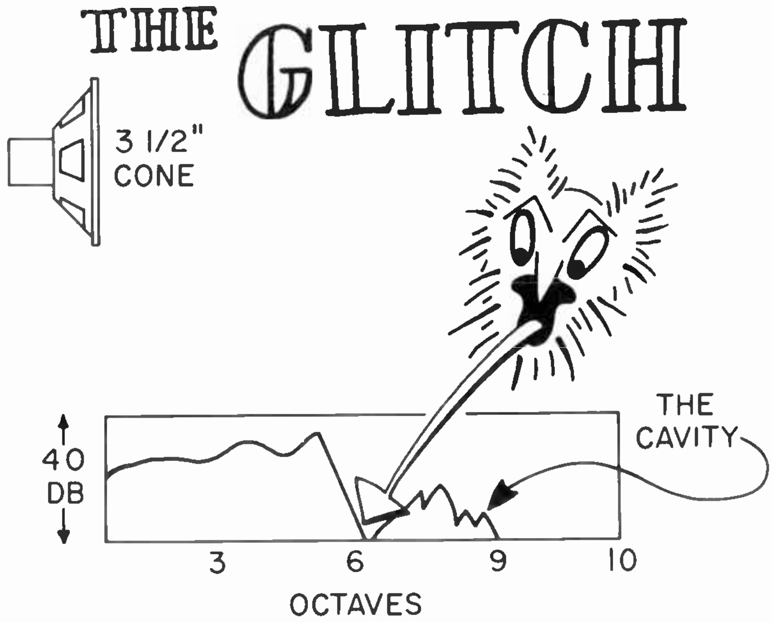


Fig. 6—The low energy output of cone type drivers is very high with a 3½-in. diameter cone, but is not easily damped. Operating as a pressure device in a sealed cavity, a violent cancellation is shown about 2000 Hz.

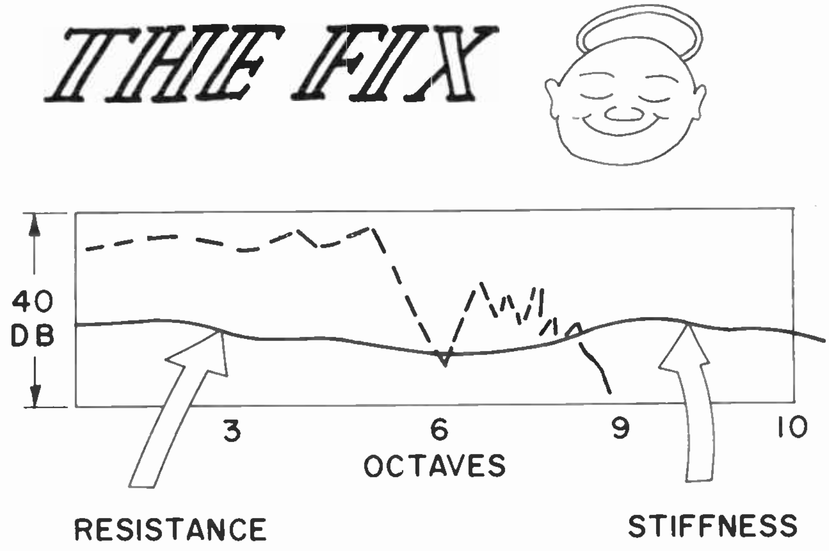


Fig. 7—With a 2-in. flat diaphragm acoustic resistance lowers the bass efficiency to the bottom of the "glitch." Increased stiffness and lower mass extends the high end.

is accomplished at the expense of the unneeded bass efficiency by applying a very high acoustical resistance to the area back of but not touching the diaphram. This works almost entirely at the low end to reduce all bass level to the bottom of the glitch, effectively banishing it. By compressing the felt used as a resistance the bass level can be adjusted for pleasing balance, or accented to appeal to the younger listeners. If physicists would only think of compromises like this there would be no need for engineers!

Exploiting the Pressure Design

The cutaway illustration of the new PRO-4AA headphone (Fig. 8) shows how the preceding considerations have resulted in a practical design which performs well. The felt washer is adjustably clamped to damp the bass range. A 1 inch very light weight self-supporting voice coil operates in a gap saturated with flux from the magnet structure. This coil is attached to a 2-mil mylar

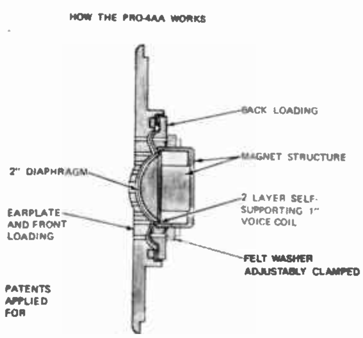


Fig. 8—Cross section of the Koss Pro—4AA with 2-in. element.

diaphragm that is actually sandwiched between 2 conforming plastic members. In this way high acoustical stiffness is achieved because the trapped air volumes are kept exceedingly small, thus giving extended 24 kHz response at full level. Note the plastic dome beneath the diaphragm which greatly minimizes the trapped air volume.

The conforming liquid filled ear cushion effects an almost perfect seal against the head with comfort, thus insuring linear bass response to the electrical limits of the supporting amplifier.

A collateral, but important benefit of the unique acoustical loading is that the plastic of the "sandwich" supports the mylar diaphragm uniformly against destructive excursions from all but the most severe overloads, making the element virtually blow-out proof.

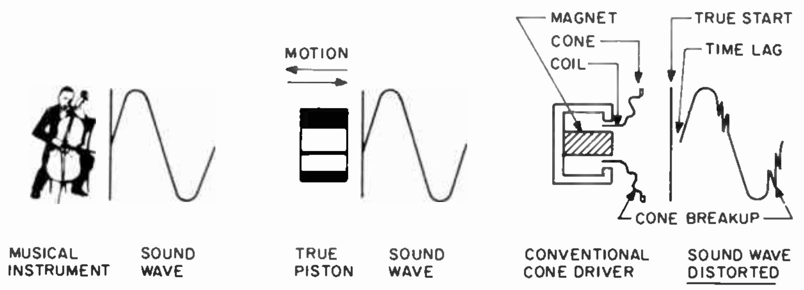


Fig. 9—Ideal piston operation and cone breakup.

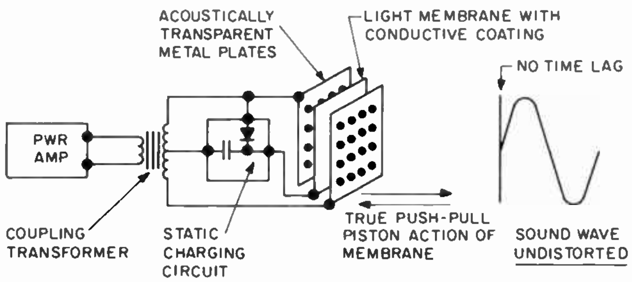


Fig. 10—Operation of the electrostatic driver. The charging circuit may be self-energized as shown, or be separately energized from a power supply, as with the Koss ESP-9.

ELECTROSTATICS

The illustration (Fig.9) shows a cone loudspeaker and an electrical signal in the form of pure sine wave. Lacking high stiffness, the cone vibrates not only at the frequency of excitation, but breaks up and vibrates at frequencies other than the fundamental. This is a form of distortion which is present in greater or lesser degree in all dynamic drivers because only the central portion of the cone adjacent to the voice coil is under perfect control.

In the search for flattest, most extended response range and lowest distortion, the electrostatic principle overcomes many disadvantages of the dynamic units and offers several important improvements.

The moving diaphragm can be very thin, say ½ mil mylar, weighing less than a ⅜ inch layer of air adjacent. While the ideal of "infinite stiffness" may seem distant, it is closer than we might first surmise.

In a dynamic driver the magnet structure provides an intense field. This field changes the condition of space around the voice coil so that current flowing through the coil causes it to move and transmit its motion, indirectly, to the diaphragm. In electrostatic transducers a static charge on the membrane also changes the condition of space. When the diaphragm is mounted between two acoustically transparent plates, and the signal is impressed across the plates, the entire diaphragm is electrically controlled, *or stiffened*, and force is exerted *directly* over the entire area, causing it to move without breakup as an ideal piston.

The resonance of the pressure operated electrostatic driver is in the region of 2 kHz because of its low mass. This means that below this point we predict, and obtain, virtually linear response. Aided by the high electrical stiffness as well as the low mass, extended high response is obtained, and attention to proper coupling to the ear results in excellent flatness. True push-pull action, not attained in dynamic designs, cancels all second harmonic distortion.

The effect of the performance of a good electrostatic headphone is immediate, vivid and compelling. The bio-acoustic benefits of very low distortion require study, but electrostatics have a cleanliness of reproduction not approached by any other form of reproducer. There is a satisfying, unstrained quality form the extended, flatter response that surveys an unparalleled emotional experience.

In electrostatic headphones we can find the happy blend of art and science. Try listening to them—and we think you'll agree that here is modern acoustical engineering in its highest, most sustained flight. Æ

(1) ***Journal of the SMPTE*** **Vol. 61, September, 1953, Physical Factors in Auditory Perspective, J.C. Stienberg and W.B. Snow.**

The "OPEN-AIRE" Principle in High Fidelity Headphones

Friedrich Warning*

During the last decade, stereo headphones have become increasingly popular. Everyone who experiences headphone sound for the first time is amazed at the almost unbelievable difference in sound quality as compared to loudspeaker reproduction.

However, early headphone designs were not without their drawbacks. Conventional units, for example, required an airtight seal between phone and ear to achieve good low-frequency reproduction, for with the slightest leak, bass tones seemed to "drain" from program content. While the seal produced some possibly desirable effects, such as isolating the listener from his surroundings, some side effects made this a mixed blessing.

For one thing, the sense of isolation, except in special applications, often proved uncomfortable. To see why, one only need hold up a glass or cup to the ear. After a relatively short period of time, the constant pressure and lack of airflow generally proves annoying. Another problem is that the seal prevents ever-present body heat and moisture, as well as bass energy, from escaping, producing a warm and often humid environment that can quickly become uncomfortable, particularly on sticky days.

The headphone construction techniques required for this type of design also had some drawbacks; as some manufacturers produced increasingly heavy and more soundproof enclosures, some headphones became rather monstrous in size and weight. For the majority of listening situations, smaller size and weight are of definite importance, particularly for prolonged periods of time.

*Sennheiser Electronic

Getting Back to Basics

Because the "open-aire" approach to headphone design depends on a number of basic transducer phenomena, it might be well to review some of the principles originally discovered in loudspeaker research. Figure 1 shows the cone of an ordinary loudspeaker suspended in free air.

Early loudspeaker designers found that, at normal listening distances (A and beyond, Figure 1A), free-standing speakers are relatively inefficient at low frequencies. This is, of course, because sound waves generated by the *rear* of the speaker arrive at the ear

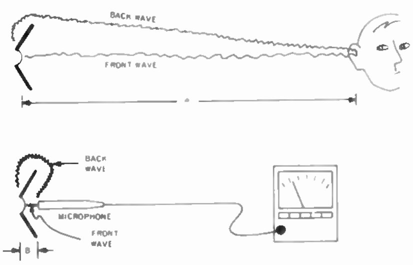


Fig. 1—Close to the speaker, phase cancellation is minimal, even at low frequencies.

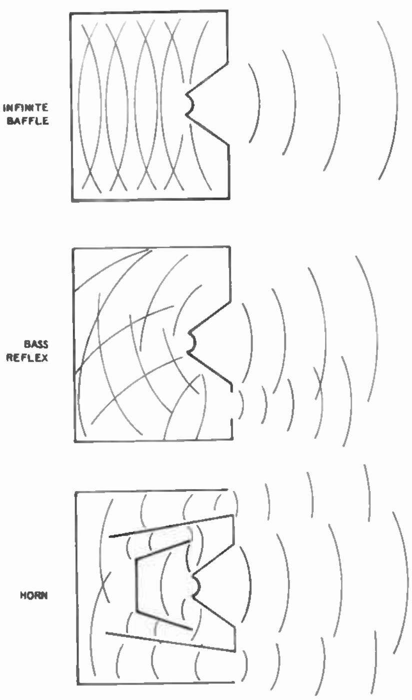


Fig. 2—Various speaker enclosures which eliminate, attenuate, or rephase back waves.

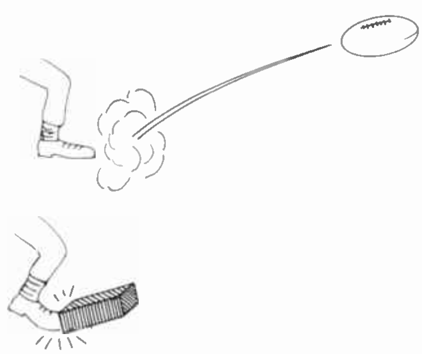


Fig. 3—An object with low mass moves easier and further when a force is applied.

with amplitudes nearly equal to those arriving from the *front* of the speaker. Since, at low frequencies, the wavelengths are relatively long, sound energy from the rear of the speaker arrives almost perfectly out of phase with energy from the front of the speaker,

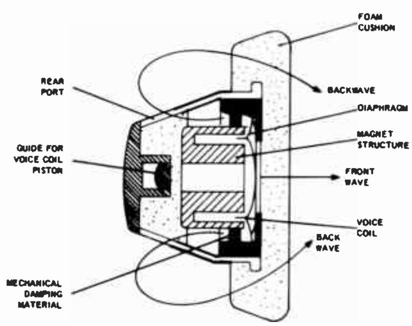


Fig. 4—Cross-section of "open-aire" headphone.

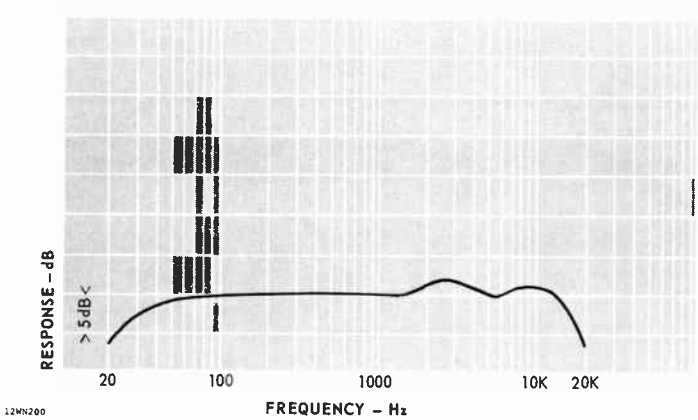


Fig. 5—Response of Sennheiser HD 414 "open-aire" headphones.

resulting in a high degree of cancellation and poor or nonexistant bass.

However, when the speaker's response is measured at distance B, which is close to the front side of the cone (actually, a distance less than the cone's diameter), much better bass response is measured. This is because, while the front wave travels a very short distance, the back wave's path is, relatively speaking, much longer. And when you consider that sound, like any form of energy, dissipates in intensity with the *square* of the distance between source and listener, the cancellation produced at B between front wave and back wave is negligible.

Of course, with a speaker system, it is impractical to move the listener to point B. To get around this problem, various types of enclosures were developed—infinite baffle, bass reflex, horn, etc. But all utilized the same basic principle, attenuate or eliminate the speaker's back wave (see Fig. 2). As the many excellent speakers on the market demonstrate, this has been successfully achieved.

But this was not done, however, without some compromise. Restricting the air-flow behind a speaker generally raises its resonance point which is not especially critical in a speaker system (since transducers may be readily made large and heavy enough to have low resonances even when enclosed), but quite another situation occurs when transducers are small, as in headphones.

When headphones are enclosed at the rear of the diaphragm, substantial increases must often be made in diaphragm size and/or mass, to reattain low resonance frequency. Unfortunately, as moving mass increases, so does inertia. This can have detrimental effects on transient response, since higher masses require greater forces to move them (or longer time to get moving with equal applied force—see Fig. 3). Thus, with conventional headphone designs, it may be necessary to trade off transient response for better bass response (or vice versa).

The "Open-Aire" Design

Going back to Fig. 1B, we see that there are few response problems if the listener is relatively close to the speaker diaphragm. In fact, if distance "B" is less than the diameter of the cone (or diaphragm), we can *ignore the nullifying effect* of the back waves for all practical purposes. And if this is achievable, there is no longer any need to "bottle up" or attenuate the back wave, with associated resonance and transient response problems.

Now consider, for a moment, the ear's location with respect to the diaphragm of a headphone. You see the point—without resorting to a huge diaphragm, the ear is still less than half-a-diameter away from the diaphragm. It is precisely this "free-air" principle that is responsible for the excellent frequency and transient response of the "open-aire" design.

Figure 4 illustrates the cross-section of an "open-aire" headphone. A moving coil system is coupled to a diaphragm in the normal way. However, the housing at the rear of the driver element is perforated, to allow unrestricted passage of the sound generated by the rear of the moving diaphragm. There is no trapped air behind the diaphragm to raise resonance or inhibit transient response. So, with a diaphragm of proper weight, elasticity, and diameter, it is possible to obtain good low-frequency response combined with excellent performance in tone-burst tests.

Ear as a Resonant Cavity

Even though "open-aire" headphones do not seal the *rear* of the transducer, they could be constructed in a manner that eliminates all outside sounds by providing the *front* of the diaphragm with an airtight coupling to the ear cavity. However, besides the comfort problems raised earlier, two acoustic problems could result.

First, a sealed cavity in *front* of the diaphragm would have precisely the same injurious effects on frequency and transient response as one behind it, raised resonance and all its results.

Second, tests have shown that several disturbing resonances can occur between the outer ear and a flat surface close to it—a phenomenon not uncommon with conventional headphones. In the "open-aire" design, a soft, lightweight cushion of porous foam spans the opening of the outer ear. The damping effect of this cellular material prevents dips and peaks in the frequency response. Figure 5 illustrates the response of a typical "open-aire" headphone, the Sennheiser Model HD 414. The intentional response rise at 2.5 KHz corresponds to the natural increase in sound pressure at this frequency caused by the dimensional properties of the head of an average person.

Psychological Factors

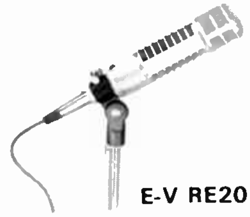
While our discussion of the "open-aire" design is primarily an acoustical one, it would be incomplete without mentioning some relevant psychological factors. As the intentional response peak mentioned above indicates, our consideration of creating "optimum" sound has gone beyond the restricted environment between diaphragm and ear because any reproducing transducer should not aim at some theoretical norm (e.g., idealized "headphone" sound, "speaker" sound, etc.), but rather at verisimilitude, a lifelike-as-possible recreation of the original performance.

Thus, we have included a response peak, to duplicate the apparent sound perceived by the *ear* in an open-air listening environment.

For similar reasons, tests have shown the "free-space" feeling of the "open-aire" design are critical to the "naturalness" of using headphones. Some subjects, in fact, reported that at mid and high frequencies, the sound actually seemed to be coming from "beyond" the headphones: a fact which later tests determined was due to the *positive* effects of miniscule. backwave leakage that occurred at these frequencies.

Also, many subjects indicated a preference for the relative lack of isolation from outside sounds provided by the headphones. Responses ranged from the practical ("I know if the phone is ringing or the baby crying") to the musically-oriented ("I feel like I'm at a live performance rather than in an artificial environment"). However, objective tests revealed that, even though the headphones cannot be "heard" well by nearby individuals (even when played at high levels), wearers could achieve a feeling of total isolation, when desired, simply by advancing the volume control. Æ


AKG 200E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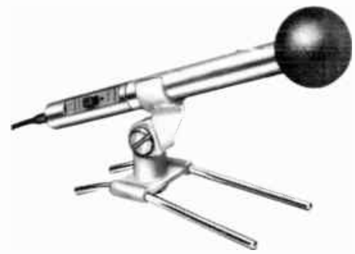
Microphones

MANUFACTURER	Model	Directional pattern	Operating principle	Case material	External finish	Impedance, ohms	Frequency response, Hz to kHz, ± ? dB	EIA Sensitivity, dBm	Mic connection	Cable length, ft.	Cable plug type	Dimensions, in.	Weight, oz.	Mounting method	Price	SPECIAL FEATURES
AKG (PHILIPS)	C-451E	Card.	Cond.	Metal	Chrome	200	20-20k	−39	XLR			5⅞ x ¾ dia.	4½	⅝-27	199.00	Mod. sys. w/interchgble. pick-up capsules.
	D-200E	Card.	Dyn.	Metal	Paint	200	30-16k	−55	XLR	15		7 9/16 x 1⅝ dia.	8	⅝-27	79.00	Two-way card. TS, high imp., $90.00.
	D-190E	Card.	Dyn.	Metal	Chrome	200	40-16k	−50	XLR	15		6¼ l. x 1½ dia.	6	⅝-27	59.00	TS, high imp., $70.00.
	D-1000E	Card.	Dyn.	Metal	Chrome	200	40-16k	−51	XLR	15		6 l. x 1[illegible] dia.	9½	⅝-27	69.00	Equal. swit. TS, high imp., $80.00.
	D-160E	Omni.	Dyn.	Metal	Chrome	200	50-16k	−55	XLR	15		5⅞ [illegible] 1⅜	7½	⅝-27	65.00	Adj. resp. curve.
ADVENT	MDC-1	Card.	Dyn.	Metal	Black	500	50-16k		Capt.	20	Phone plug	5½ x 1¼	10	Clamp spld.	90.00 pr.	Mtchd. pair W. carrying case, desk stands.
ASTATIC	810	Card.	Dyn.	zinc	Satin chrome	150 40k	40-15k	−147	Sw. A4M	18	None	1 5/16 dia x 5 9/16 long	8	Adapt. to ⅝-27	$85.00	Switch version, 810S. Also avail. gold plated.
	820	Omni.	Dyn.	Steel	Satin chrome	150 40k	40-18k	−154 −158	fixed	18	None	¾ dia. x 9 long	6	adapt to ⅝-27	$79.50	Switch version, 820S. Also avail. gold plated.
	840	Omni.	Dyn.	Alum.	Satin chrome	150	50-12k and 50-16k	−153	fixed	30	None	¾ dia. x 2⅝ long	1¼	neck cord or tie clip	$85.00	Lavalier Type. Extra-flexible cable. Dual Response choice. Switch version 840S. Also avail. gold plated.
	857L	Card.	Dyn.	Zinc	Satin nickel	150	50-15k	−150	Sw. A3M	18	None	2 1/16 dia. x 6⅛ long	8	adapt. to ⅝-27	$66.00	Ball-head filtering. Switch versions 857H & 857H-S.
BEYER (REVOX)	M550S	Omni	Dyn.	Metal	Matte blk	200 to	70-18k ±3	−152	Cap.	7	Jack	5 x 1 dia.	8	Clamp inc.	35.00	
	M810	Card.	Dyn.	Metal	Matte blk	200 to 5k	50-16k ±3	−148	Can	16	Jack	5 x 1¼ dia.	9	Clamp inc	57.50	
	Soundstar XIN	Card.	Dyn.	Metal & plast.	Matte dk grey	200	30-18k ±2	−146	Can	16	Open end	7 x 1¼ dia.	11	Clamp inc	70.00	
	M69	Card.	Dyn.	Metal	Matte blk	200	50-16k ±3	−144	Can	16	Open end	7 x 1 dia.	8	Clamp inc	85.00	
	M101	Omni	Dyn.	Metal	Matte blk	200	40-20k ±2	−150	Can.	16	Open end	4½ x ⅞ dia.	3½	Clamp inc.	105.00	
	M67	Card.	Dyn.	Metal	Matte blk.	200	40-18k ±3	148	Can.	16	Open end	8 x 1½ dia.	11	Clamp inc.	110.00	
	M201	Spr. Card.	Dyn.	Metal	Matte blk.	200	40-18k ±2.5	149	Can.	16	Open end	6 x [illegible] dia.	8	Clamp inc.	165.00	
	M88	Spr. Card.	Dyn.	Metal	Matte blk.	200	30-20k ±2	144	Can.	16	Open end	7 x 2 dia.	9	Clamp inc.	200.00	
ELECTRO-VOICE	631A	Omni.	Dyn.	Zinc	Satin chrome	HiZ or LoZ*	80-13k	HiZ: −151 LoZ: −149	A3M to A3F	15	Not furn.	6 x 1⅜	6	Stand clamp	38.70	Sealed mag. on/off swit. w/removable outside actuator; *specify.
	1710	Omni.	Elect. cond.	Alum.	Bge. anod. w/enam. trim	150	80-13k	−142	Integ. cable	18	Mini plug	8¼ x ⅞	9	Stand clamp	39.75	On/off swit.
	1750	Card.	Elect. cond.	Alum.	Bge. anod. w/enam.	150	80-13k	−137	Integ. cable	18	Mini plug	8¼ x ⅞	9	Stand clamp	45.00	Same as above.
	670V	Card. single "D"	Dyn.	Alum. alloy	"Top Brass"	LoZ or HiZ *	60-14k	LoZ: −150 HiZ: −152	A3M to A3F	15	Not furn.	7¼ x 1½	6	Stand clamp	54.00	Vol. contl.; integ. "pop" filter. *Select. (Specify; tie clip supplied.
	624	Omni.	Dyn.	Diecast metal	Matte gray	HiZ or LoZ *	100-7k	−152	Integ. cable	18	Not furn.	3 1/4 x 1¼	14	Lav. neck cord	31.50	*Specify; tie clip supplied.
	RE55	Omni.	Dyn.	Steel	Bge. Mico-matte	150	40-20k	−149	A3M to A3F	18	Not furn.	10½ x ¾	8½	Stand clamp	162.00	Stand or hand held; metal carrying case incl.
	RE20	Card.	Dyn.	Steel	Bge. Mico-matte	50, 100, 150	45-18k	−150	A3M to A3F	18	Not Furn.	8½ x 2⅛	26	Stand adapt.	285.00	Cont. Var. D® design; built-in pop filter; bass tilt-down swit.
	DL42	Card.-line	Dyn.	Alum. & steel	Bge. Mico-matte	150	50-12k	−144	A3M to A3F	Not furn.	A3M	16¾ x 3¾	13	Shock mnt. w/boom adapt.	330.00	Long-reach line mic; boom, fishpole or handheld; wind-screen, handle and carrying case incl.

Microphones

Hammond M-100

Lafayette Cardioid



Mura DX-285

PML EC-71

MANUFACTURER	MODEL	Directional pattern	Operating principle	Case material	External finish	Impedance, ohms	Frequency response, Hz to kHz, ± ? dB	EIA Sensitivity, dBm	Mic connection	Cable length, ft.	Cable plug type	Dimensions, in.	Weight, oz	Mounting method	Price	SPECIAL FEATURES
EMPIRE	TM-6		Dyn.	Alum.	Alum.	250	50-15k	0.1 mV		13	Furn.	7¼ x 1¼	5.7	Tbl. stnd.	49.80	
	T-M5	Omni	Dyn.	Alum.	Alum.	600	70-13k		Can. XLR3	15	Can.	6⅝ x 1⅜		Tbl. stnd.	89.80	
HAMMOND (MICROSOUND)	M-100	Omni	Cond.	Alum.	Anod.	50-200	20-20k ± 3	−157	Can. XLR	15	Pin & Phone	4 x 1 dia.	3	⅝-27 w/adapt.	269. * 159. **	*Stereo pr., ** mono; Nuvistor preamp; a.c. power supply.
LAFAYETTE	Omni Elect. Cond.	Omni	Elect. Cond.			600	20-13k			10	¼" std.	6 x ⅝	10	Desk stand	18.95	Foam windscreen, batt. inc., FET crty.
	Card. Dual Imp.	Card.	Dyn.	Diecast	plshd. chrome	600/50k	100-10k			20		8 x 1	10	Swiv. adapt.	19.95	Wiremesh grille.
	Card Elect. Cond.	Card.	Elect. Cond.			600	30-16k			20	¼" std.	10½ x ⅞	24	Desk stand	29.95	Foam windscreen; batt. incl.; FET crty.
	Deluxe Ball	Omni	Dyn.	Diecast	Satin alum.	250/50k	100-10k			6		6¼ x 2⅛	30	⅝-27	18.50	Ball screen; blk. metal desk stand.
MAGNAVOX	1A9226	Omni	Cond.	Brass	Nickel	400	50-15k +12, −6			20	3.5 mm MinPH	7¼ x ⅞		½" pipe	29.95	Stand incl.; batt. in mic case.
	1A9212	Card.	Dyn.	Brass	Nickel	50k 600	300-10k +8 −6			20	3.5 mm MinPH	6⅜ x ¾		½" pipe	29.95	Incl. stand.
	1A9211	Omni	Dyn.	Diecast zinc	Black	50k	80-10k ± 6			10	3.5 mm MinPH	6¼ x 1		½" pipe	14.95	As above.
	1A9210	Omni	Dyn.	Plastic	Black	50k	100-10k ± 6			6	3.5 mm MinPH	6 x 1			9.95	As above.
MURA	DX285	Omni	Elec. cond.	Alum.	Alum.	600	30-18k			20	¼" Phone	6 x ⅜ dia.	5	Stand	29.95	
	DX-129	Card.	Dyn.	Anod. Alum.	Blk. & alum.	600; 50k	40-14k	−58	COAX	20		8 x 1 dia.	8	⅝-27 stand adapt.	29.95	
	WX-172	Omni	FM wire-less	Alum.	Alum. & blk.		100-7k					1' x 1½" x 6"		⅝-27 stand adapt.	39.95	
OLSON	EC 100	Card	Elect	Alum	Brshd gold	low	30-16k ± 1.5	140	A3S	20	not furn.	8½ x 2½	5	clamp	36.00	
	EO 200	Omni	Elect	Alum.	Brshd gold	low	30-16k ± 1.5	135	A3S	20	not furn	8½ x 2½	5	clamp	39.60	
	MM-327	Card	Dyn	Alum	Brshd. gold	600, 50k	50-15k ± 2	125	Amph	20	not furn.	7½ x 1⅞	10	clamp	22.98	On-off swit.
PML (ERCONA)	EC71	Card.	Cond.	Metal	Satin chrome	30-50; 200; 600; HiZ	40-18k, ± 3	−164	Preh plug	12	None	2⅝ x ⅝ dia.	1¼	⅝ x 27 stand adapt	109.50	Micro min. cond. mic.; power supply a.c. or d.c.
	EK71	Omni	Cond.	Metal	Satin chrome	30-50; 200; 600; HiZ	40-18k ± 3	−164	Preh plug	12	None	2⅝ x ⅝ dia.	1¼	⅝ x 27 w/stand adapt	99.50	Micro min. cond. mic; power supply a.c. or d.c.
	D44	Card.	Dyn.	Metal	Blk. chrome pltd. grid	200	60-16k	−165	att.	12	None	5	4.7	⅝ x 27 w/stand adapt	34.95	Avail. w/ on/off swit.; 30 ft. cable 200 ohms bal. or Hi-Z at 39.95.
	TC4 US-V	Card. Omni. Bi.	Cond.	Metal	Anti-refl. satin chrome	50 or 200	30-20k	−172	Tuchel att.	20	None	5⅝ x 1 1/16 dia.	5	⅝ x 27 w/stand adapt	350.00	Studio FET mic; power supply a.c.

Microphones

Radio Shack Cardioid

Sennheiser MNH-415

Shure 585SA

MANUFACTURER	MODEL	Directional pattern	Operating principle	Case material	External finish	Impedance, ohms	Frequency response, Hz to kHz, ± 3 dB	EIA Sensitivity, dBm	Mic connection	Cable length, ft.	Cable plug type	Dimensions, in.	Weight, oz.	Mounting method	Price	SPECIAL FEATURES
PML (HERVIC)	DC96	Card.	Cond.		Brshd. chrome	200	30-18k	−61	Can.	15	Not furn			Std. adapt. & sk. mts.	242.95	
	TC4V	Var.	Cond.		Brshd. chrome	200	30-18k	−56	Can.	15	Not furn			Std. adapt. & sk. mts.	279.95	
	DC-63	Var.	Cond.		Brshd. chrome	200	25-18k	−60	Can.	15	Not furn			Std. adapt. & sk. mts.	413.95	
	EK71	Omni	Cond.		Brshd. chrome	HiZ	80-18k	−38	Oir	15	Not furn			Std. adapt. & sk. mts.	97.95	EC71 same but cardioid, $97.95.
	DC73	Card.	Cond.		Brshd. chrome	200	30-20k	−60	Can.	15	Not furn			Std. adapt. & sk. mts.	160.95	
RADIO SHACK	1045	Card.	Elect. cond.	Die-cast		600; 20k	30-15k		Can.	15					32.95	W. pop screen, desk std.
	1044	Omni	Elect. cond.	Die-cast		600; 20k	30-15k		Can.	15					27.95	As above.
	Highball	Card.	Dyn.	Die-cast		50-250; 50k	50-15k		MC1F	15					39.95	Dual-Z; w. ¼ in. phone plug adptr.
	Highball 5	Card.	Dyn.	Die-cast		600; 20k	70-13k		¼ in. phone	15					32.95	Dual-Z.
SENNHEISER	MD441	Sup. Card.	Dyn.	Metal	Chrome leather	200	40-20k ± 2	−145.8	XLR			10.6 x 1.4 x 1.3	16	⅜"	236.00	Adj. freq. resp.
	MD421	Card.	Dyn.	Metal/plast.	Non-refl. flat blk.	200	45-17k ± 5	−145.8	XLR			7 x 1.8 x 1.7	14	⅜"	154.00	
	MKH815	Beam	Cond.	Metal	Chrome	20	60-20k ± 4	−115	XLR			22 x ¾ dia.	14	Clamp	599.00	
	MKH415	Card.	Cond.	Metal	Chrome	20	40-20k ± 3	−121	XLR			10 x ¾ dia.	6.1	Clamp	499.00	
	MKH435	Sup. Card.	Cond.	Metal	Chrome	20	40-20k ± 2	−121	XLR			7½ x ¾ dia.	5	Clamp	470.00	Studio music mic.
SHURE	585SA	Card.	Dyn.	Diecast zinc	Chrome	HiZ	50-13k	153½	Amph. MC1F	15	Not furn	6¾ x 2⅛	13½	Adapt.	47.25	Unisphere A; also avail. LoZ model 585SB.
	588SA	Card.	Dyn.	Diecast zinc	Chrome	HiZ	80-13k	155	XLR	15	Not furn	6½ x 2⅛	12	Adapt.	40.80	Unisphere B; also avail. LoZ model 588SB.
	515SA	Card.	Dyn.	Diecast zinc	Blk. & chrome	HiZ	80-13k	154		15	Not furn	6½ x 1½	12	Adapt	28.20	Unidyne B; also avail. LoZ model 515SB
	579SB	Omni.	Dyn.	Diecast zinc	Chrome	LoZ	50-15k	151	XLR	20	Not furn	6⅜ x 1½	5½	Adapt.	47.25	Vocal sphere.
	548	Card.	Dyn.	Diecast zinc	Blk. & chrome	HiZ; LoZ	40-15k	151	XLR	15	Not furn	6⅛ x 1½	9	Adapt.	72.00	Unidyne IV; also avail. w/ mag. reed swit. as 548SD.
	565	Card.	Dyn.	Diecast zinc	Blk. & chrome	HiZ; LoZ	50-15k	150½	Amph MC4M	15	Not furn	6 x 2	11	Adapt.	67.80	Unisphere I; also w/mag reed swit. as 565SD.
	545	Card.	Dyn.	Diecast zinc	Blk. & chrome	HiZ; LoZ	50-15k	151	Amph MC4M	15	Not furn	5¾ x 1¼	9	Adapt.	60.00	Unidyne III; also w/mag. reed swit. as 545SD.
	55SW	Card.	Dyn.	Diecast zinc	Chrome	HiZ; MedZ; LoZ	50-15k	151½	Amph MC3M	15	Not furn	7¾ x 3⅛	26	⅝ x 27	61.20	Unidyne II w/ on/off swit
	SM7	Card.	Dyn.	Diecast zinc	Gray enam.	150	40-16k	150	XLR	Not furn	Not furn	7½ x 3¾	27	⅝ x 27	240.00	Resp. tailoring sys.
	SM33	Super Card.	Ribbon	Diecast zinc	Gray enam.	30-50; 150-250	40-15k	148	XLR	20	Not furn	8 x 1¼ x 1⅞	26	⅝ x 27	156.60	

Microphones

Sony ECM-22

Turner 45

Unicord CE-2

MANUFACTURER	MODEL	Directional pattern	Operating principle	Case material	External finish	Impedance, ohms	Frequency response, Hz to kHz, ± ? dB	EIA Sensitivity, dB m	Mic connection	Cable length, ft.	Cable plug type	Dimensions, in.	Weight, oz	Mounting method	Price	SPECIAL FEATURES
SONY	ECM-250	Card.	Elect. Cond.	Alum.	Satin	200	50-14k	57		16½	Mini	6 x 1½	1.3	Hand	49.95	FET elect.; int. batt. oper.
	F-98	Card.	Dyn.	Alum.	Bge.	Lo	70-14k	58		6½	Mini	6¼ x 1⅛	6½	Hand	12.95	
	EOM-51	Dmni	Elect. Cond.	Alum.	Blk. & satin	50; 250; 600	50-16k	53.2		10	Cann. XLR-3	7¾* x 1¼*	4	Hand	149.95	*Dut to 17½"; multi. imp.; telescopic capsule.
	ECM 54P	Card.	Elect. Cond.	Alum.	Satin	250	20-20k	53.8	Cann.	20	Pigtail	7⅜ x 1	6.35	Hand	149.95	1 low cut filt.; cond. or elect. cond. oper.
	ECM-18	Card.	Elect. Cond.	Alum.	Satin	250	50-12k	56.8		6½	Mini	6½ x ¾	5.3	Hand	22.95	
	ECM-21	Card.	Elect. Cond.	Alum.	Satin	50: 250: 600	40-16k	53.8		18	Pigtail	6¾ x ¾	8	Hand	59.95	
	ECM-22P	Card.	Elect. Cond.	Alum.	Satin	250: 600	40-15k	54.8	Cann. XLR	20	Pigtail	7⅝ x 1	4	Hand	99.95	Low cut filt.; phantom pwr. oper.; multi. imp., swit.
	ECM-95-S	Card.	Elect. Cond.	Alum.	Satin	1500*	70-10k	50		4½	Mini & sub-mini	5½ x 1⅛ x 1	5	Hand	19.95	*May be term. w/ lower imp. load.
	ECM 280	Card. uni.	Elect. Cond.	Alum.	Satin	200	30-18k	56		16.5	Pigtail	6 x 1	5.1	Hand	79.95	Low cut filts.; int. batt. oper.
	ECM 220	Card. uni.	Elect. Cond.	Alum.	Satin	200; 10k	50-20k	56/57		16.5	¼" phone	7.28 x 1.45	9.6	Hand	39.95	Mic holder suppl.; multi. imp. (Hi-Lo).
	ECM 16	Dmni	Elect.	Alum Cond	satin	600	50-13k	57.8		6	Mini	1½ x [illegible]	1.09	Lapel	29.95	Int. batt. supply; FET elect.
TEAC	MC-105	Uni	Dyn.		Chrome	10k	50-15k			5½	Phone plug	5¾ x 1⅛			55.00	Can be conv. to 600 ohms.
	MC-106L	Omni	Dyn.		Black	10k	50-15k			5½	Phone plug	5½ x 1			20.00	Same as above.
	MC-201	Uni	Cond.		Chrome	10k	50-15k			9	Phone plug	6 x ⅞			80.00	Same as above.
TURNER	35	Omni	Dyn.	Alum.	Gold paint	150; 25k	50-12k	−154 −156	Wired in	25	Not furn	3 x ¾ dia.	1¾	lav. clip	70.00	Dual impedance.
	500	Card.	Dyn.	Diecast zinc alloy	Satin chrome	dual 150 40k	40-15k	151	A4F	20	Not furn	6⅞ x 1½ dia.	12	⅝-27 thrd hldr	100.00	Specify S500 for rotary type on-off swit.
	600	Card.	Dyn.	Diecast zinc alloy	Satin chrome w/blk. frnt.	40k	50-15k	151	Amph Mc2m	20	Not furn	6 x 1⅝ dia.	14	⅝-27 hldr	70.00	Model 602 same w/imp. of 150 ohms, $70.00
	45	Card.	Dyn.	Blk. cycolac w/alum. head		hi	100-13k	151	Perm att	20	Phone plug	7 x 1¼ dia. caper.	6	⅝-27	45.00	
	2300	Omni	Dyn.	Steel	Satin chrome	40k	50-15k	151	A3F	20	Not furn	6 x 1¼ dia.	8	⅝-27 thrd hldr	80.00	Model 2302 same w/imp. of 150 ohms, $ [illegible] 00.
UNICORD	CE-1	Card.	Elect Cond.	Anod Alum.	Satin	600	40-18k ± 5	−45	Con.	20	Tel. plug	8¼ x 1⅝ dia.	7		42.95	MT-1 Trans. avail. (plug-in) type for 50k output.
	CE-2	Card.	Elect. Cond.	Anod. Alum.	Satin	600 Bal.	40-18k	−45	Can.	20	Tel. plug	8¼ x 1⅝ dia.	7		55.00	CE-1, CE-2 use FET amp built-in w/1.5v pen cell.

The Case for the Condenser Microphone

A report by Richard Fowle*

"Incorporation of the electret capsule and the resultant increase in battery life, makes the use of superior condenser microphones as convenient as the use of dynamic microphones."

ACCURACY is the key factor in the design and performance of any audio component. There are many ways of measuring the deviation from absolute accuracy. *Total harmonic distortion, intermodulation distortion, frequency response, signal-to-noise ratio, phase response, impulse response,* and many more are terms describing the relative accuracy of a component. The aim of all these measurements is to show how closely the output of a device approximates the input to that device.

In studying the important characteristics of condenser and dynamic microphones, it quickly becomes apparent that a properly designed condenser microphone is *inherently more accurate* and thus better than a properly designed dynamic microphone.

A microphone is a device which converts acoustic energy into electrical energy. In other words, when sound of a given frequency and amplitude strikes the diaphragm of a microphone, alternating electrical current of equivalent frequency and amplitude is produced by the microphone. This transformation takes place in several well ordered steps, regardless of the type of microphone.

1. Acoustic energy (an alternating air pressure) strikes the diaphragm of the microphone.
2. The acoustic energy becomes mechanical energy as the diaphragm vibrates in accordance with the difference in pressure between front and rear sides of the diaphragm.
3. The mechanical energy (vibration) of the diaphragm is converted to electrical energy (alternating current) in accordance with the intensity and frequency of the sound pressure.

From the above, it is apparent that the distortion may first occur in step 2, where a diaphragm is required to react with extreme accuracy to a constantly varying sound pressure.

The diaphragm of a condenser microphone is a circular piece of extremely thin (typically 0.00025 in. thick) plastic or metal which is supported at its edge. (See Fig. 1-A). The diaphragm of a dynamic microphone is also a thin plastic or metal sheet supported at its edge. The diaphragm of the dynamic microphone is connected at its center to

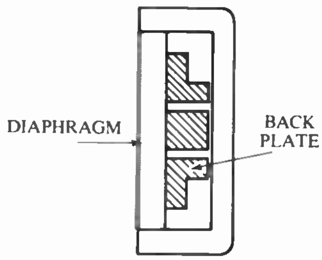


Fig. 1-A—Cross-section of a condenser microphone.

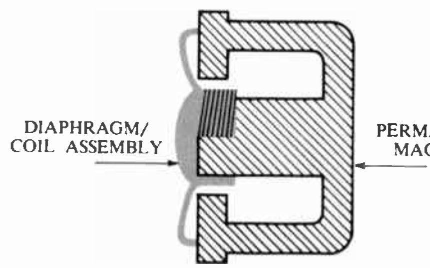


Fig. 1-B—Cross-section of a dynamic microphone.

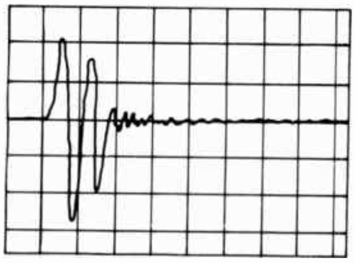


Fig. 2-A—Impulse response of a Sony condenser microphone.

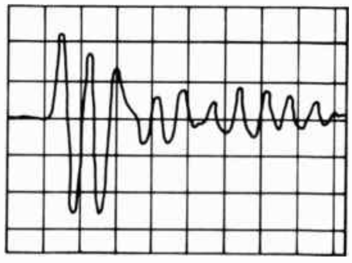


Fig. 2-B—Impulse response of a professional dynamic microphone.

*Sony Product Manager, Superscope, Inc.

a coil of wire which moves in a magnetic field whenever sound pressure strikes the diaphragm. (See Fig. 1-B).

The coil attached to the diaphragm of the dynamic microphone is required to convert the mechanical vibration of the diaphragm into electrical current, whereas the condenser microphone does not require a coil. By adding enormously to the mass of the vibrating system, this coil prevents the dynamic microphone from responding accurately to variations in sound energy. [1]

A simple experiment can be performed to illustrate and verify this effect. Two microphones are placed side by side, a $50 Sony condenser and a well-known $150 dynamic. A spark gap (as in an automobile spark plug) is used to produce a sound impulse. The output of both microphones is displayed on an oscilloscope. (See Figs. 2-A and 2-B.)

Once set in motion by the impulse, the high mass of the dynamic microphone diaphragm causes it to continue in motion (and thus produce output) despite the complete absence of sound. The low-mass diaphragm of the Sony condenser microphone ceases to move as soon as the sound stops. From this experiment it can be deduced that whenever there is a change in sound pressure, either in amplitude or frequency, the condenser microphone will respond quickly and accurately to the change while the dynamic microphone will adjust to the change more slowly.

The higher mass of the dynamic microphone's moving system also creates other problems, the most important of which is resonance. Any object possesses one or more resonances, as determined by the object's mass and other factors. Generally speaking, the larger the mass, the lower the resonance. [2]

In a microphone, the output will increase sharply at the resonant frequency of the diaphragm. Ideally, the resonance of a microphone diaphragm should be well above the audio frequency range in order to avoid an audibly peaked output. Only the very finest (and most costly) dynamic microphones have resonances restricted to the frequency range above 15 kHz because of the inherent high mass of their diaphragms. In contrast, the resonance of the low-mass diaphragm of a condenser microphone will be at an extremely high frequency, resulting in smooth, peak-free response throughout the audio range. [3]

The characteristics of condenser microphones will give the sound a natural quality which is unattainable with any but the most expensive dynamic microphones. Furthermore, in public address applications where feedback is a problem, a dynamic microphone will often cause feedback at its resonant frequency, thus reducing the maximum volume capability of the system. The smooth response of a condenser microphone will generally permit substantially higher volume levels before feedback occurs.

The low-mass diaphragm of a condenser microphone provides many advantages relative to the dynamic system. The sensitive condenser diaphragm will produce less harmonic and intermodulation distortion at a wider range of frequencies than the dynamic diaphragm. The condenser diaphragm is less sensitive to low frequency mechanical vibration transmitted through the stand and microphone case to the diaphragm. Finally, condenser microphones, with built-in pre-amps, generally have a higher output level than dynamic microphones. Therefore, the condenser microphone will produce an acceptable signal-to-noise ratio, even when used with less than ideal microphone preamplifiers. (See Figs. 3-A and 3-B.)

[1] As an analogy, take a baseball player and two bats. One bat weights 38 ozs., the other weighs only 28 ozs. The player steps to the plate with the heavier bat. The first pitch looks good so the player starts to swing the bat. In the middle of his swing, he realizes that the pitch is not a strike, so he attempts to stop his swing, but the inertia of the heavy bat causes it to continue forward, and a strike is called. The player then switches to the lighter bat and the same situation reoccurs. This time, as soon as the player attempts to stop his swing, the bat stops. In this example, it can be seen that the higher the mass of a moving object, the more the object resists a change in motion.

[2] As an example, take an empty 16 oz. glass and an empty 4 oz. glass. Strike both with a spoon. The larger, more massive glass will resonate at a lower frequency than the small glass.

[3] Although the problem of diaphragm resonance differs in cardioid microphones and omni-directional microphones, it is generally true that the frequency response curves of dynamic microphones show more peaks and dips as well as narrower bandwidth than those of condenser microphones because of the low frequency resonance of the dynamic microphone's diaphragm and its associated acoustic circuit. A detailed explanation of the factors involved is too complex to present here. Please see references 1 to 4 for information on this subject.

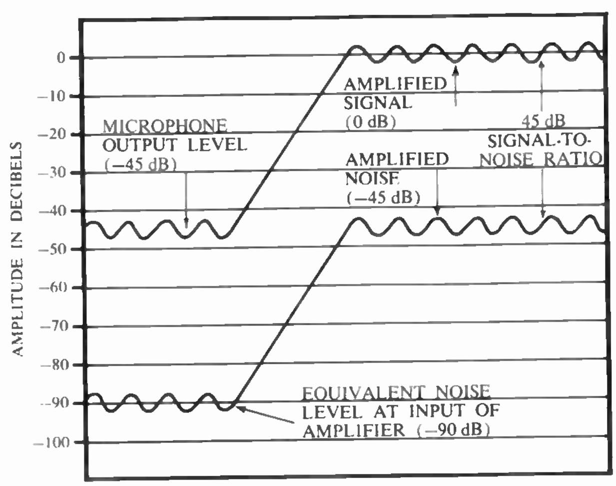


Fig. 3-A—Signal-to-noise ratio of a condenser microphone with an output level of -45 dB when used with a microphone pre-amplifier having a high noise level.

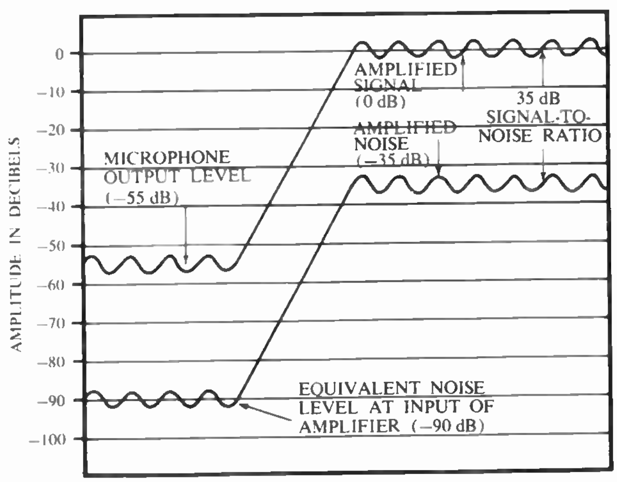


Fig. 3-B—Signal-to-noise ratio of a dynamic microphone with an output level of -55 dB when used with a microphone pre-amplifier having a high noise level.

After reviewing the preceding facts, which clearly indicate that the condenser microphone is technically superior to the dynamic microphone, a natural question is "Why do dynamic microphones out-sell condenser microphones by a margin of at least 10 to 1?" Up until now, three negative characteristics of condenser microphones have made these technically superior products unusable for all but the most professional applications.

First, since the condenser microphone is an electronic device (each contains an amplifier), a power source is required for it to operate. Originally, the amplifier used a vacuum tube and the microphone required three different voltages: 4 to 12 volts for the filament, B+ for the plate, and 64 to 200 volts to polarize the condenser capsule. As a result, a bulky, complex, and costly external power supply was required, with a huge multiconductor cable interconnecting the power supply and the microphone. See photos.

With the introduction of transistors, the power requirements were simplified, but the condenser microphone still required two voltages, 1.5 to 12 volts for the transistor amplifier, and 65 to 200 volts to polarize the capsule. A few

Sony C-37A and power supply (1955)

Sony C-37FET (1964)

condenser microphones with self-contained battery power supplies appeared on the market, but battery life was still comparatively short. The majority of condenser microphones still required an external power supply. The microphones were still too complex for general purpose use.

In 1969, Sony Corporation manufactured the first electret condenser microphone, the ECM-22. The incorporation of an electret capsule [4] further reduces the power requirements. A single voltage, from 1.5 to 9 volts, is required. Since no high voltage is required for polarizing the capsule, battery life is extended (up to 1100 hours in the ECM-22; up to 15,000 hours in other Sony condenser microphones).

Each of these technical advances reduced the complexity of the condenser microphone, thus making it usable for a wider variety of purposes. The latest advance, the incorporation of the electret capsule and the resultant increase in battery life, makes the use of superior condenser microphones as convenient as the use of dynamic microphones.

Secondly, because the condenser microphone is very complex, it was also extremely susceptible to damage from shock, moisture, humidity, and heat. The reduced complexity of the modern condenser microphone design has resulted in a dramatic improvement in durability. Sony condenser microphones, for example, are extremely rugged and will withstand without any ill effects the normal accidents which occur, as for example, falling off a table. However, these condenser microphones cannot be used to hammer nails (a capability one well-known manufacturer of dynamic microphones claims for his product), since they are precision audio components and must be treated as such.

Finally, prior to 1969, the least expensive high quality condenser microphone cost over $200 with the majority of such microphones costing from $275 to $500 each. At these prices, the only users of condenser microphones were professional recording engineers, whose income depended entirely on the quality of their recordings, and acoustical engineers, who used condenser microphones for audio measurements which required greater precision and quality than any available dynamic microphone could provide. After designing the electret condenser microphone, Sony manufacturing engineers were faced with the task of producing these microphones at prices which were competitive with dynamic microphones. They were able to meet this challenge, and as a result, Sony condenser microphones are available at prices starting below $20.00. Each is a true condenser microphone, and thus incorporates all of the inherent advantages of even the most expensive condenser types. Æ

Sony ECM-22

[4] A discussion of electret capsules, or of operating principles of condenser microphones, would be too complex to present here. Please see references 5 to 7 for information on these subjects.

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MICROPHONES-- Quo Vadis?

James H. Kogen*

A Comparison of the Dynamic and Electret Condenser Transducers for Sound Reinforcement Microphone Applications

ONE NEED NOT be very sophisticated in the art of microphone design to discover that there is no panacea for all problems. Of necessity, any practical design is to some degree a compromise. The best microphones are those in which the most important characteristics have been optimized. What constitutes the "most important characteristics" will differ depending upon the application. A microphone that is optimum for one application may not be so for another. To make a fair and honest comparison of microphones, we must consider the application and base our analysis on those characteristics that are significant for the situation being studied.

The selection of a microphone for a given application is ultimately made by the user. The engineer designing a microphone can only anticipate which features will be most important in determining that selection. His choice of features must be based on a thorough analysis of the many design variables and possible compromises. This requires a careful evaluation of all the pertinent characteristics of available transducer types and acoustical designs, consideration of special features, and a thorough study of the many techniques of mechanical construction. Ultimately, his design will be based upon what he conceives to be the best combination of all of these characteristics to satisfy the requirements of the intended application.

A comparison between the electret condenser transducer and the dynamic transducer must take into account many factors. In this article we will discuss these factors, particularly as they apply to microphones for use in sound reinforcement. Emphasis will be placed on identifying the important factors by which a comparison of transducer types can be made. The conclusions as to which microphone is most advantageous for a given situation will be left to the reader.

The electret condenser transducer is an entirely feasible device for use in microphones. This transducer must, however, be judged by the same standards applied to dynamic, ribbon, ceramic, and other types of transducers commonly used in microphones. As stated previously, no transducer offers a panacea. The successful use of the electret condenser transducer will depend upon its particular features and limitations and the way in which these relate to the application of the microphone.

A comparison of the electret condenser transducer and the dynamic transducer, in general purpose sound-reinforcement microphones, covers a very broad range of applications. We will not at this time consider applications such as professional recording and broadcast, mobile communications, and laboratory test microphones. Clearly, the conclusion reached with regard to sound-reinforcement application might not apply for these other applications since the emphasis may be on different characteristics.

Complexity of Microphone Evaluation

When one considers the details, the evaluation of a microphone is a complex matter, and this is particularly true for directional microphones. There is no simple overall criterion of performance that describes the quality of a microphone; there are many criteria for making a judgment and each must be considered separately. Consider the following characteristics by which a microphone can be judged.

1. *Sound Quality.* Several factors affect the sound quality of a microphone. These factors include the frequency response, the polar response (the relative sensitivity of the microphone in all directions), and the distortion at all sound-pressure levels, from the minimum to the maximum to which the microphone will be subjected. Although these parameters can all be measured with great accuracy, our ability to relate these parameters in detail to the subjective sound quality is still quite limited. As a result, in addition to making the laboratory measurements, we must also make an evaluation based on listening—which of course is entirely subjective.

2. *Extraneous Noises.* Since microphones are designed to respond to minute changes in sound pressure measured in microbars (a microbar is one-millionth of barometric pressure or 14.7 x 10^{-6} psi), they are often sensitive to other kinds of mechanical energy input as well. Structure-borne noise can be very disturbing in many applications. When a microphone is held in the hand, for example, a variety of characteristics become important, such as cable noise, frictional noise caused by rubbing the hand or clothing against the microphone, and "thump" noise when the microphone is placed on a floor stand. Such noise can be a very significant factor in judging the quality of a microphone.

Another type of extraneous noise is the "pop" that often occurs when a user expresses the letter "p" or "t." In close-talking applications, excessive "pop" sensitivity can make a microphone practically unusable.

A third type of extraneous noise is that produced by wind. In outdoor applications, the relative sensitivity of a microphone to wind noise may well determine whether or not the microphone can be used.

3. *Reliability.* A microphone with a multitude of superb features, but with poor reliability, is essentially worthless. Sound-reinforcement applications require reliability often under conditions of severe abuse. We have seen microphones swung by their cables and dropped on floors on many occasions. To qualify as a reliable sound-reinforcement product, a microphone should be capable of being dropped on a hardwood stage without deterioration of performance.

Other factors related to reliability are humidity and temperature. Sound-reinforcement microphones are employed outdoors in sub-zero weather and in the heat of a tropical sun. They are used in arid desert regions as well as in highly humid atmospheres.

4. *Output Level and Signal-to-Noise Ratio.* Output level is a significant factor because the signal-to-noise ratio of the system depends upon the output of the microphone in relation to the noise of the system (usually as determined by the input

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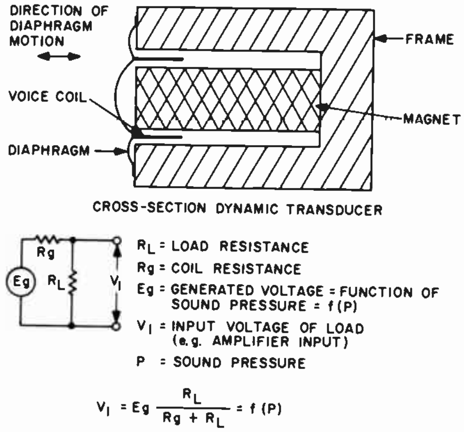

Fig. 1—Equivalent electrical circuit of a dynamic transducer.

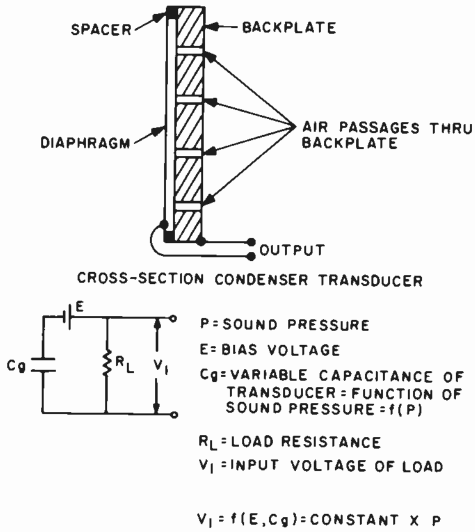

Fig. 2—Equivalent electrical circuit of a condenser transducer.

stages of the mixer or preamplifier). A higher output level from the microphone can be advantageous in improving the signal-to-noise ratio. However, an excessively high output level may overload the input stages of the mixer or preamplifier. This means that the output level must be designed to consider peripheral equipment as well as the internal design of the microphone itself.

These are just a few of the factors that must be considered in evaluating a microphone. Each factor must be considered in detail, and specifications, where possible, must be assigned to assure proper performance. The selection of a transducer must then be made in terms of how that transducer performs for each and every one of these specifications. As we stated initially, no transducer offers a panacea. Each transducer has its own particular features and we must select the device that best suits our application requirements.

In the remainder of the article, we will first describe the operating principles of the dynamic and the electret condenser transducers, and then we will compare the two on the basis of a number of important specifications. These specifications include power supply requirements, frequency response, polar response, handling noise, "pop" and wind noise, reliability, output level, distortion, and transient response.

Operating Principles and Major Characteristics

The following is a very brief review of the principles of operation and major characteristics of the dynamic and electret condenser transducers.

The dynamic transducer operates as an electrical generator. A coil of wire (the voice coil) is attached to a metal or plastic diaphragm that moves in response to an input of sound energy. The coil is placed in a magnetic field and a voltage is produced when there is relative motion between the coil and the magnetic field. The dynamic transducer is a self-generating device that requires no external source of power. The equivalent circuit and the equation that relates output to input, shown in Fig. 1, is highly simplified and presented to indicate the steady-state relationship between sound input and electrical output.

The three significant characteristics of the dynamic transducer pertinent to much of the discussion later in this article are:

1. It is self-generating and requires no external power supply.
2. It has a low internal impedance in the range of 25-1000 ohms at all frequencies in the audio spectrum.
3. As compared to the condenser transducer, it has relatively high diaphragm-coil mass.

Both the standard and electret-type condenser transducers convert acoustical energy into a variation in electrical capacitance. This variation occurs when the diaphragm is moved by a sound pressure, thus changing the distance between the diaphragm and the backplate. This capacitance change is reflected as an electrical output in a circuit, such as that shown in Fig. 2. The microphone acts as a varying series element in this circuit. (As in Fig. 1 for the dynamic transducer, this circuit is simplified in order to show the relationship between the acoustical input and the electrical output.) The equation relating output voltage to acoustical input is also shown in Fig. 2. In an electret condenser transducer, the bias voltage results from a permanently stored electrical charge in the transducer; conventional condenser transducers require an external voltage supply.

Characteristics of the electret condenser transducer pertinent to later discussion are:

1. It is a self-generating device requiring no external power supply.
2. It has a very high impedance since it is a capacitor of a few hundred picofarads minimum and requires a preamplifier located physically close to the transducer.
3. It has minimum mass for a diaphragm-type transducer in that nothing is suspended from the diaphragm.

Comparison of Transducers

1. *Power Supply Requirements.* While both the electret condenser and the dynamic transducer are self-generating devices, there is a considerable difference in application in that the former is a high-impedance device and the latter has a relatively low impedance. It is standard practice to employ low-impedance dynamic microphones with cables of hundreds of feet in length without significant problems with hum pickup and deterioration in frequency response. This is accomplished without preamplification at the microphone location.

On the other hand, the high impedance of the condenser microphone necessitates the use of a preamplifier in close proximity to the transducer element. In practical application, the best solution is to build the preamplifier into the microphone and provide either a battery or an external power supply to energize this amplifier.

An external power supply provides a suitable solution but does mean that this extra element must be included in the system or, alternatively, d.c. voltage must be made available at the input terminals of the microphone amplifier. The latter arrangement provides the neatest solution, but at the present time, sound-reinforcement equipment is not normally provided with such a voltage source. As a consequence, if the extra element (the external power supply) is to be avoided, an internal power source must be provided, which means a battery in the microphone.

When a microphone incorporates a battery, one must immediately be concerned with battery life. It is possible to design a preamplifier with current drain so low that the battery

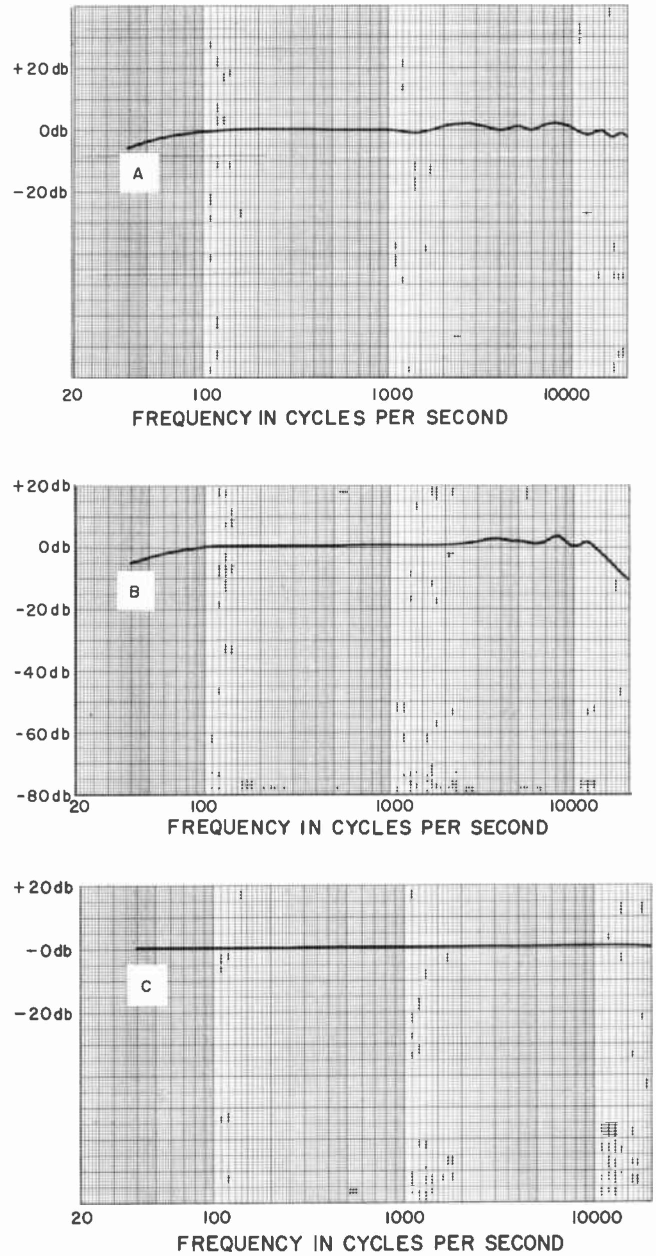

Fig. **3A**—Free field axial response of a professional quality omnidirectional dynamic microphone. **B,** Free field axial response of a professional quality omnidirectional condenser microphone. **C,** Free field axial response of a ½-in. laboratory condenser microphone.

life will be quite long. The life of a battery may vary over a considerable range, depending upon the normal quality variations of batteries and upon the temperatures to which they are subjected. The life of a battery might be as high as 10,000 hours or a little over one year. However, the variation in life is very large and might extend from as little as a few months to many years.

The major point of consideration here is that although battery life can be reasonably long, the batterv is still a replacement item that must be maintained. A dead battery means a dead microphone. A weak battery may mean a marginally operating microphone. A leaky battery could mean a damaged microphone. The dynamic microphone does not have this problem. The electret condenser transducer must offer features to overcome this disadvantage when compared to the dynamic transducer.

2. *Frequency Response.* As we have stated, the frequency response of a microphone presents only a rough indication of the sound quality. It is nevertheless a standard by which microphones are compared and must be considered as a very important basis of comparison. In the following, we will consider low frequency response extension, high frequency response extension, and mid-frequency smoothness.

In terms of low frequency response extension, the condenser transducer potentially has an advantage. Through the use of very high impedance preamplifiers, this transducer can be made to operate at frequencies well below the low end of the audio range (20 Hz). Dynamic transducers can be made to operate at very low frequencies also, but in order to achieve such response, a compromise must be made that tends to make the microphone more sensitive to handling noise. The advantage of response below 50-100 Hz is of questionable value in sound reinforcement, although this potential advantage of the condenser element might be useful in other applications.

In terms of high frequency response extension, we cannot state with certainty at this time that either transducer has an advantage insofar as audio frequencies are concerned. Figures 3A and 3B are response curves of two popular omnidirectional dynamic and condenser microphones. Both of these types exhibit response to 20 kHz within a few decibels. Figure 3C is the response of a laboratory-type condenser microphone with frequency response extending well beyond 20 kHz. For sound-reinforcement applications, both types of transducers have the capability of satisfactory high-frequency response.

Smoothness of response in the mid-region can be accomplished by both transducers, as shown in Figure 3. The condenser element might have an advantage in having fewer small variations in its response curve, but the variations in a good dynamic microphone response would be in the order of 2 dB or less, and it is doubtful whether this would affect the sound quality sufficiently to be detected when the two types of transducers are compared subjectively.

In our judgment the frequency response possibilities of the two types of transducers, for use in sound reinforcement, are similar. We must, however, remind the reader of the point made previously with regard to frequency response and subjective sound quality. Frequency response does not tell the whole story with regard to sound quality. The frequency response of a dynamic and electret condenser microphone may be similar, but the sound quality could differ because of other factors.

3. *Polar Response.* Theoretical analysis indicates that neither the dynamic nor the electret condenser transducer has an advantage with regard to polar response. This characteristic is primarily a function of the acoustical design of the microphone in conjunction with the transducer. Measurements on existing microphones have corroborated this theoretical analysis.

The unidirectional dynamic microphone requires a mass-controlled transducer having a relatively low fundamental resonant frequency. The condenser microphone requires a resistance-controlled transducer having a resonant frequency in the mid-range. The acoustical networks required to achieve unidirectional characteristics can be similar for the two transducers, and there are a variety of networks available for either type. Ultimately, we may discover that a particular network in conjunction with one or the other of the transducers offers some practical advantage. At the present time, however, this is not the case, and the two types of transducers are comparable with regard to polar response of the microphone in which they are employed.

4. *Handling Noise.* This is a characteristic that is often overlooked but one that can be very important. A microphone is

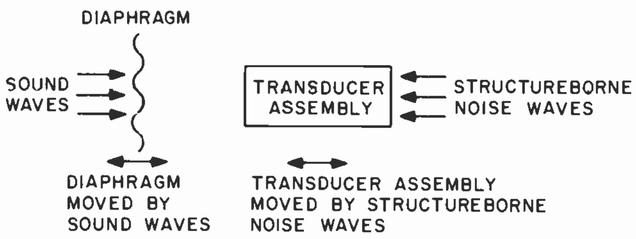

Fig. 4—Voltage generation caused by structure-borne noise. Relative motion between diaphragm and assembly produces electrical output and can result from either diaphragm or assembly motion.

a unique instrument in that it must be highly sensitive to the input of sound energy but should also be insensitive to the input of structure-borne energy. These are decidedly conflicting requirements. As a general rule, the sensitivity to structure-borne sounds will increase with the mass of the diaphragm in the transducer. The electret condenser transducer unquestionably has an advantage over the dynamic transducer in this regard. Figure 4 shows a simplified representation of how the microphone transducer reacts to structure-borne noise. In order to reduce the effects of this type of noise, it is standard practice in dynamic microphone designs to introduce a shock absorber between the outer case of the microphone and the dynamic transducer. Because of its inherent low sensitivity to structure-borne noise, the requirement for shock isolation of the condenser microphone is significantly less than that of a dynamic microphone. This results in two important considerations.

A. For a given structure-borne noise sensitivity, the cost of the shock isolation in a dynamic microphone will be higher than that of a comparable electret condenser microphone.

B. The requirement for shock isolation will add to the size of the dynamic microphone. Stated conversely, the electret condenser microphone could be made smaller because of the simplicity of the shock isolation required. This feature has merit, for example, in a lavalier microphone or a microphone mounted on a headset boom. The advantage is less in situations where microphone size is determined by other factors such as windscreens, pop filters, and cable connectors.

We feel that insensitivity to structure-borne noise can be the major advantage of electret condenser microphones in sound-reinforcement applications.

5. *Pop and Wind Noise.* Measurements on a large variety of dynamic and condenser microphones indicate that both transducers are equally susceptible to pop and wind. A reduction of pop and wind noise must be achieved through the use of external windscreens and pop filters in either case.

6. *Reliability.* As stated previously, reliability is an extremely important factor in comparing products for sound-reinforcement applications. We will consider three factors relating to reliability: mechanical ruggedness, the effect of humidity, and the effect of temperature.

With regard to ruggedness, the dynamic microphone has proved itself over a period of more than 30 years. A properly constructed dynamic microphone is sufficiently rugged to withstand the rigors of severe sound-reinforcement applications. The electret condenser microphone has yet to prove itself. The answer to the question of relative ruggedness will only be gained through experience over an extended period of time.

In humid conditions, the dynamic transducer presents no problem. Care must be taken, of course, to adequately protect metallic parts, but this is standard practice in all quality microphones. The electret transducer, on the other hand, has potentially severe problems in humid atmospheric conditions. High humidity can cause the loss of the electret charge. This was a problem in electret devices made several decades ago, using Carnauba wax, and presents a potential hazard to modern-day electrets employing plastic materials. Clearly, for the microphone to be satisfactorily reliable, the electret charge must be maintained under the extreme conditions often found in sound-reinforcement applications.

In laboratory tests, our company considers life of 1,000 hours at 100% relative humidity to be a minimum requirement for any microphone. We have tested several electret microphones that would not withstand this test at room temperature. Our conclusion is that suitable electrets can be made, but that humidity still presents a potentially serious problem for the electret.

In comparing the two transducer types under conditions of high temperature, we find the onus again is on the condenser to match the known performance capability of the dynamic. Dynamic microphones made by Shure are required to withstand storage temperatures from -20 degrees F to +165 degrees F and must operate within standard performance specification at temperatures from -20 degrees F to +140 degrees F. We have found these to be suitable temperatures to guarantee reliability in performance of the microphone under field conditions. The electret condenser microphone must, of course, also be capable of withstanding these extremes of temperature. The high end of the temperature range will offer the most difficult problem for the electret, particularly when combined with high humidity.

An additional factor that must be considered in evaluating the electret condenser microphone at high temperature is the effect of temperature on the dry cell incorporated in many of these microphones. Most alkaline and carbon zinc batteries will not withstand a temperature of +165 degrees F for an extended period of time. We would include the dry cell in temperature tests to determine whether any leakage of the cell might damage the microphone. Since the dry cell is normally easy to replace, we feel that it is reasonable to change to a new dry cell for subsequent testing after the high temperature exposure is completed.

7. *Output Level.* Since the electret condenser microphone must be supplied with a built-in preamplifier, there is a possibility of providing a very high output level—much higher than with the unamplified dynamic microphone. Care must be taken in providing a low-noise preamplifier in order to achieve suitable signal-to-noise performance. The design must also consider the problem of amplifier saturation in order to minimize distortion at high sound-pressure levels. The advantage of a high-output level is somewhat mitigated by the fact that an excessively high-output level can result in overloading the input stages of the mixer or amplifier to which the microphone is connected. Typical output of a dynamic microphone is in the order of -57 dB with reference to one volt per microbar. We would question the value of an output level of greater than roughly -52 dB because of the probability of overload in subsequent stages of amplification.

8. *Distortion.* Dynamic microphones usually exhibit very low distortion. We have measured a large variety of both omnidirectional and unidirectional dynamic microphones at levels up to 150 dB sound-pressure level. Total harmonic distortion is typically below 1% up to the highest pressure measured. (As a point of reference, 130 dB sound-pressure level can cause physical damage to the ears.) Similar measurements on condenser microphones indicate total harmonic distortion below 2% for the majority of microphones, and below 1% for the higher quality condenser microphones up to sound pressure levels of 130 dB SPL. Condenser microphones exhibit a relatively sharp overload point in the range of 130 to 150 dB SPL, at which level the distortion rises very rapidly. This type of distortion can be caused by bottoming of

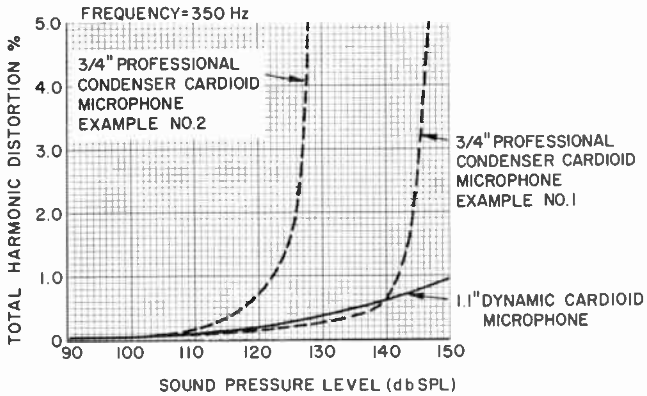

Fig. 5—Overload characteristic examples of condenser and dynamic microphones.

the diaphragm but is normally the result of clipping in the preamplifier. Figure 5 shows total harmonic distortion versus sound-pressure level for typical dynamic and condenser microphones.

The two transducers can be comparable in terms of distortion at normal sound-pressure levels and can be suitable for maximum sound-pressure levels normally found in sound-reinforcement applications. The dynamic transducer is inherently less difficult to control with regard to distortion and has an advantage in being able to handle extremely high peaks of sound pressure, which may sometimes occur in "close talk" applications.

9. *Transient Response.* At this time, there is no standard test for transient response of a microphone. While this type of test is commonly used in evaluating amplifiers, loudspeakers, servomechanisms, and so on, there are several problems in applying the test to microphones. One problem is that of creating a standard transient; and at this date, no such standard has been devised or specified. A second problem is correlating the results of such a test with other types of measurements (such as frequency response) and with subjective reaction.

Since no standard test procedure exists, and since there is little documentation with regard to the significance of transient response tests, we do not feel that it is proper to make a comparison of microphones on this basis at this time. While one might easily design a test that will display differences between two microphones, to be fair one would have to document the significance of the differences noted. One major aspect of this documentation would certainly be the subjective differences. We strongly emphasize that this type of testing must be very carefully controlled with many variables to be considered. One cannot make a judgment based on a simple demonstration.

Conclusion

In summary, then, we would like to make the following points. Any comparison of microphone types must consider the application for which the microphone was intended. Comparison of microphone types is complex and must include all of the many pertinent characteristics. In comparing transducer types, one must consider the way in which the total microphone is designed and built. Either the dynamic or the electret condenser transducer can be employed in a good or bad microphone design.

This article has attempted to describe some of the more important characteristics pertinent to electret condenser and dynamic microphones in sound-reinforcement applications. The selection of a microphone will be made by the user and will be based on those characteristics that are most significant to the application. Æ

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