THE B.A.S. SPEAKER

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In This Issue

The latest issue of the new, lo-cal Speaker contains an interesting variety of ingredients. The feature articles cover the two ends of the audio chain. The first is a piece, reprinted from Recording engineer/producer, by audio consultant F. Alton Everest on comb filtering in microphones and rooms. It should offer valuable insight into a little-known effect for all you devoted recordists, and may help explain why some multi-miked productions sound so bad. The second is a thorough analysis of speaker cables, both garden-variety and exotic, which further strengthens the case for thick wire of ordinary construction. New contributors Charles Ward, James Thompson, and Mallory Harling are to be commended for a solid piece of work.

Local member Mark Fishman has an interesting tale to tell about the time he tried to find out where Dolby level really is on a cassette; the answer turns out to be a little more complicated than you might think. An elegant summary of recent issues of Hi-Fi Choice comes from Jack Reed. We'll be hearing from Jack on more technical matters in upcoming issues.

The summary of the March BAS meeting, at which Gotham Audio's Stephen Temmer held forth on digital recording and other topics, will hold the interest of those who remember his role in the digital standards controversy of a year ago. Temmer' has seen and heard just about every digital audio and video storage system now extant, and is able to put these experiences into the context of his thirty-odd years in the business.

Call Up Your Friends

By the time you read this, the annual BAS telephone list will be available. Send a SASE to Frank Farlow at Box 7 for your copy.

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For Sale

- Verion/Cotter MK-1 step-up transformer for moving coil cartridges, "P" strapping for low-impedance/low-output cartridges (e.g., Supex, Fidelity Research, Ortofon, Koetsu, Signet), mint, $275. R. Luppen, 10309 Seabury Lane, Los Angeles, CA 90024 or call (213) 581-8121 days, (213) 474-8138 evenings.
- 1 : 1 dubs of audiophile master tapes made by Monmouth Recording Co., cassette or reel-to-reel, dbx II or Dolby. Send SASE for details to Ronald Freeman, 43 Stratford Dr., Freehold, NJ 07728 or call (201) 462-5855 after 7 p.m. EDT.
- Quad ESL, old (#5546) but in first-class condition both sonically and visually, $275. Francis Daniel, 201 West 89th St., New York, NY 10024 or call (212) 874-0590.
- ADS Model 10 Acoustic Dimension Synthesizer, includes power amp and speakers, $695; SWTP Tigersaurus Amp, 200 watts mono, and 198 preamp, kits partially built but never finished, both for $75; Dynavector DV505 arm, $225; Shure V15-III with both spherical and elliptical styli, used only a few hours, $35; Shure V15-IV with spherical (3 hours) and elliptical (10 hours) styli, $75. Doug Robinson, 104 Lincoln, Waterloo, NB 68069 or call (402) 779-2589.
- Crown D-150A Amp and IC-150 preamp, both for $550; Onkyo TX4055 tuner, $125; Sherwood SEL-300 tuner with digital display, $250; Advent 201 cassette deck, just reconditioned at factory, $200; SWTP Armadillo amp, $100; stereo electronic crossover, 2 or 3-way, 900 Hz and 6,000 Hz, 24 dB/octave, $100. Michael Marks, (617) 732-4855 days, (617) 469-0710 evenings.
- Recording equipment: 3 Sony ECM-2'2P and 1 Sony C-55 FET condenser microphones, custom mixer (6 in, 2 or 4 out, ghosted center channel or panning, very low noise, VU meters, Beyer transformers), assorted bronze braided cables, 3 floor stands (9 ft), 4 desk stands, carrying case, all for $450. Magazines: Journal of the Audio Eng. Soc. 1961-1979 complete except 3 issues, Audio Amateur 1973-1979 complete, BAS Speaker Vols. 3, 5, 6, 7 complete, Audio magazine 1973-1979 partial (70 issues), all for $75. Ron Roscoe, 62 Harris St., Acton, MA 01720 or call (617) 263-8296 evenings, (617) 493-6084 days.
- Van Alstine Double Dyna 400 with external power supply box, excellent condition, $700; Dynaco CAB-2D vented cabinet, excellent condition, $15; Infinity Black Widow tone arm, GF arm tube and fluid damper with new container of damping fluid, good condition, $160; Russound tape switching unit with directions, unused, mint, $20. Chuck, (215) 623-0752 evenings, Fridays, or weekends.
- Dyna PAT-5 with Jung (and other) mods, $150; GAS Sleeping Beauty Shibata, $75; SME III tone-arm with spare carrier, $150; Blaupunkt Frankfurt AM/FM stereo auto radio, $100; Jensen 4-in. coax auto speakers, $20 the pair; PS Audio PS-III phono section, accepts MM or MC cartridge, $125; Quatre DLH-100 power amp (not the one that blows up), separate supplies and other mods, 100 watts/channel, $175. All equipment functionally and cosmetically excellent. Julian Vrieslander, 364 Trumbull Corners Rd., Newfield, NY 14867 or call (607) 256-3838 weekdays.
- Selling my "mint" audio system due to home computer upgrading expenses: Infinity 4.5 RS speakers, new 2/80, $2,800; Symmetry ACS-2 crossover, new 4/80, make offer; Audio Research D-52B amps, 6/79 and 2/80, $775 each; Audio Research SP-6B preamp, modified 4/80, $950; Powerlite MC-3 head amp, rare, $250; Koetsu MC1 moving coil cartridge, new 2/80, $500; Linn Ittok LVII arm, 2/80, $475; Linn LP-12 turntable, 9/78, $625; Yamaha T1 tuner, 4/80, $275; Nakamichi 700 cassette deck, 2/77, good condition, $475; Stax SR5 headphones with STD7 transformer, 2/77, excellent condition, $110. Will consider trades plus cash, but only serious inquiries, please. Gerry, (408) 738-3310 evenings (PDT).
- RTR ESR-15 electrostatic tweeters with built-in crossover at 1,200 Hz, $500/pair; special design, limited production 435 watts (rms) mono tube amplifier, only the very best parts used, $500. Terry Eckert, New York, (212) 226-0199.

Wanted

- Altec "Flamenco" equipment cabinet. Will pay 10% finder's fee for information leading to purchase. David Klein, 25 Trinity Street, Danvers, MA 01923 or call (617) 774-8434.

Corrections

Lux Loudness

In a review of the Lux 5C50 preamp in the October/November issue, Norm Relich praised the unit for the behavior of its loudness control, saying that the change in sound is gratifyingly small over the range of available volume levels (p. 9). This prompted an editor's note that mistakenly assumed the existence of a separate control labeled "loudness." In fact there is no such control on the 5C50; Mr. Relich was referring to the preamp's apparent ability to sound the same over a wide range of volume control settings without the aid of a conventional "loudness" circuit to boost levels at the frequency extremes. He concludes from this that the need for such controls has been exaggerated.

Turntable Tests

My Yamaha YP-D8 turntable and Signet TK-7E cartridge comprised sample number 22 in the BAS turntable clinic held March 25, 1979.

In part 1 of the review of the clinic results published in the January 1980 Speaker, paragraph six states that sample number 22 was a moving coil cartridge. Table 2 on page 32 also indicates, via the parentheses around the total capacitance used during the frequency response test, that the cartridge was a moving coil type. It was not. The Signet TK-7E is a moving magnet cartridge.

Also in error are the figures shown for the capacitance setting of the Apt preamp and the total arm + cable + preamp capacitance used during the frequency response plot.

My data sheet from the clinic shows that the preamp capacitance was set at 50 pF, and I remember choosing this value because most published reports on this cartridge seem to agree that 270 pF produced flattest response. Thus, the total capacitance used with the Signet TK-7E during this test was 289 pF.

-- Bill Asher

Dolby Level: Take your pick

Dolby level for cassettes is specified as 200 nanowebers/meter (nWb/m) flux density. It has also been measured as -1 dB referred to DIN 0; DIN 0 is specified as 250 nWb/m. It is intuitively obvious to the most casual but sophisticated observer that 200 nWb/m and 250 nWb/m are 2 dB apart. What is going on? Just what is Dolby level, anyway? And why do we care so long as it is always the same?

My interest in this started because I own several different cassette decks and was uncertain of their relative Dolby tracking. Were tapes made on one machine being properly decoded by the others ?

Dolby level is important because it establishes a relationship between a specific flux level on the tape and a specific voltage level in the deck electronics. The proper decoding of Dolby-encoded tapes requires that this relationship be maintained for all recordings of any origin. However

When I inquired about the metering on a CTF900, U.S. Pioneer informed me that they place DIN 0 (250 nWb/m, remember) at +4 on their meters and that their "0" is at Philips reference level, 160 nWb/m. But they put Dolby level at +3 dBl Furthermore, they claim that Dolby level is actually 224.6 nWb/m, using as their rationale that, "200 nWb/m of surface magnetism, the Dolby specification, is inaccurate because a tape head does not read only the surface magnetism." Hmmm.

So is Dolby level always the same? Further inquiries were directed to Dolby Labs and two prominent manufacturers of test tapes, BASF, representing the German-European viewpoint, and TDK, representing the Japanese, who manufacture most of the cassette decks used in the USA. BASF, as expected, uses DIN standards for almost everything and provided the following explana--tion:

There are two methods of flux measurement in general use. They are of equal relative accuracy but produce numbers which differ about 0.8 dB. The Dolby specification was written using the method developed by J. McKnight, formerly of Ampex Corporation. This involves a playback head of specific design and materials. The DIN standard, however, specifies a different measurement method for determining its specified reference level, and the DIN method yields higher numbers for the same piece of tape.

Dolby Labs points out that confusion has arisen because the reference level, which has not changed and which "has been very accurately maintained by comparison through time, is now measurable in two ways, each developed to reduce confusion.

TDK's engineer sent a long letter intended to be helpful, but it was not entirely successful. This was one occasion when the language barrier, Japanese /English, prevented a clear trans-mission of the message. After describing in some detail the method used by TDK for measuring flux density of a constant level recording, the letter points out that the two measurement methods are accurate to approximately only 0.25 dB anyway. Unfortunately, he goes on to claim that Dolby level is -1.8 dB re DIN 0, which contradicts the claims of U.S. Pioneer, BASF, and Dolby Labs.

Sony Corporation has an interesting-and question-begging-approach to resolving the issue. On some of their latest cassette decks with LCD metering, Dolby level is marked at -1.5 dB re DIN 0, or about halfway between the two possible levels. Any mistracking due to a +0.5-dB mismatch on someone else's cassette is probably inaudible or masked by other frequency response errors, so this is a practical "solution."

My own approach is probably theoretically preferable: I assume (1) that Dolby Labs knows whereof it speaks regarding Dolby level and that (2) BASF is in better command of English than the Japanese. Therefore, I have chosen -1 dB re DIN 0 (or +3 dB re Philips reference level) as representing the most probable Dolby level and have simply calibrated all my various cassette decks this way. All of my tapes are now interchangeable, and so far I have not noticed any sonic anomalies on prerecorded cassettes.

- Mark Fishman (Massachusetts)

Hi-Fi Choice(s), 1980

Those familiar with the British Hi-Fi Choice series of equipment reviews, which are revised at intervals of 1½ to 2 years, may be interested to know the planned publication dates for new editions this year, which I recently received from the publisher in response to my inquiry:

"Cassette Decks and Tapes"	Available
"Turntables and Tonearms"	April 7
"Receivers"	August
"Cartridges and Headphones"	October

Evidently, the old editions of "Tuners" and "Amplifiers" are to continue mostly unrevised for a while. The new volume on "Stereo Systems" covering "over thirty" of these ready-mounted component systems which appeared last year and "Loudspeakers 3" continue also, of course.

Those unfamiliar with the series may be interested in some description and summary. I have only the "Turntables and Tonearms 2" and "Cartridges and Headphones" volumes (Hi-Fi Choice, Nos. 12 and 13, respectively), both copyright 1978 and both written by Martin Colloms. Some other volumes are written by Angus MacKenzie.

The guides are about 210 digest-size pages long including some advertising, and they include indexes, five-to-fifteen-page introductions describing what the component does and how it should be used as well as how the measurements and listening tests were carried out, recommendations, and comparison tables. Each review includes a photo of the component taken for Hi-Fi Choice, an article of about 340 words discussing compatibility with other components, test results, listening quality, a table of "General Data" mostly giving about thirty measurement results, and a couple of pen recorder or oscilloscope photo representations of further tests.

In the case of cartridges, some points covered include body mass, estimated compliance, low-frequency resonance in the SME 3009111 arm (undamped), sensitivity, load recommendations (not necessarily the manufacturers'), recommended arm mass and damping, hum sensitivity, high-frequency resonance, tracking ability at 300 Hz and 10 kHz. Frequency response and cross-talk to 40 kHz are graphed and 1-kHz square-wave response oscillograms are also included. Subjective sound quality is rated on a seven-value scale from "adequate" to "excellent" based on "blind" auditions which include disc versus master tape comparisons. The five-member jury listens over KEF R105's in a room with a reverberation time of 0.3 second, give or take 20% over the frequency range.

In a field of 81 cartridges from thirty manufacturers, ten finished in the "excellent" sound-quality category characterized by "open, neutral, deep, precise imaging." The list includes:

ADC XLM III, but vague stylus profile, mild veiling, some edge added to complex passages;

Audio Technica Signet Mk. IIIE-less apparent distortion than usual for a moving coil, a trifle bright with touch of surface noise, +2 dB above 8 kHz;

B&O MMC 20CL-distant;

Entre 1-required 2.5 grams for 300 Hz at +18 dB, hardening of sound and high-frequency grit on complex passages;

JVC X-2-slight shift toward hardness and brightness on louder, complex passages;

Mission 773-required 2.5 grams for 300 Hz at +18 dB, disc noise slightly emphasized with trace of grit, fizz and sibilance exaggeration, some poor tracking on heavy bass transients;

Ortofon VMS 20E II-slightly nasal and dull, poor tip polish, specimens vary;

Ortofon M20 FL Super-slightly obtrusive surface noise, some marginally cold steely quality;

Supex SD 900E Super-suspicion of occasional mistracking, 2.7 grams required for 300 Hz at +18 dB, slurred sibilants, high treble fizz, grain, and emphasis; and the

Ultimo 20A-2.2 grams required for the 300-Hz passage, hint of grittiness occasionally, 47-K and 68-nF load improves sound quality.

The sometimes well-regarded Ortofon MC 20 was ranked "average" on account of apparent loss of bass, and a hint of harshness and muddying of detail, especially on louder passages. The Shure V15-V was "below average" because its sound lacked depth and hardened on voice.

In the case of turntables and pickup arms, some points covered include speed drift under load, called "dynamic wow," effective arm mass, provisions for arm adjustment, and cueing drift. In addition there are the usual efforts to assay wow, flutter, rumble, pivot friction, and offset geometry. Arm resonances are measured with a small accelerometer one-third of the way down the arm from the pivot. The turntables are also checked for microphony of the turntable-arm combination in an equalized sound field and for mechanical shock sensitivity. A subjective assessment of overall sound quality is made using mainly Supex 900E Super cartridges.

In a field of 75 turntables, one, the Linn Sondek LP12, ranks "very good" in overall sound quality and eleven ranked "good." These are the B&O 1902, 2200, and 4002, the Jonorhurst JBE Series II, Micro Seiki DDX 1000, Monitor Audio ET 500, Revox B790, Sony PSX7, Strathclyde STD 305D, Technics SL 100011, and Thorens TD 126111, assuming good choice of pickup arm in each case where this is not already included.

In a field of eighteen separate arms, one, the Mission 774, ranks "excellent" and three "very good"-the ADC LMF 1 (higher frequencies subdued), Grace G707 (slightly bright and coarse), and SME 3009111 (slight loss of definition in bass and midrange), assuming good choice of turntable.

Headphones were tested for frequency response on a Neumann KU80 dummy head with a Bruel and Kjaer 4153 artificial ear. They were tried by a panel of six for sound quality and comfort. Of forty headphones from nineteen manufacturers, seven received the top sound-quality rating of "very good." The list:

B&O U70	"Slight veiling"
Beyer ET 1000	"Bass power restricted, bright, average comfort"
Leak 3000	"Slightly bright, below average comfort"
Sony ECR400	"Slight midrange weakness, below average comfort"
Stax SR40	"Lighter bass, moderately brash and overbright"
Stax SRE	"Mild fizz in high treble, slight midrange weakness"
Yamaha HP-1	"Some midrange hardness"

This sampling should give a fair indication of the breadth and depth of the series and may even enable you to compare it with your own experience. Except for the two old editions, which cost £ 1.50 each, the price is £2.00, plus 50p each for postage, payable by bank draft in English pounds. The books may be ordered from the publisher, Sportscene Publishers Ltd., 14 Rathbone Place, London W1P 1DE, England. Mine took a month to arrive.

Jack Reed (Illinois)

OR

BAS members may also obtain recent issues in the Hi-Fi Choice series from John Tooley, RFD 2, Box 120E, Milton, DE 19968. John's rate on the issues he has in stock is $6 U.S. for book rate mailings and $7 First Class. At present he has a few copies of "Stereo Systems," "Cassette Decks and Tapes 4," "Cartridges and Headphones," and "Loudspeakers 3." What John doesn't have, he will endeavor to obtain and spare you the hassle of currency conversion. Local members may obtain the last two titles directly from Peter Mitchell at local meetings while the stock lasts and save postage altogether.

- Henry Belot

In the Literature

Audio, April 1980

- Tape Guide (p. 6): Demagnetizing may not be needed.
- Audio ETC (p. 8): On some 48-year old stereo recordings.
- Behind the Scenes (p. 18): On some of the most advanced new products.
- Trends for the Future (p. 28): Surveying new products at the Winter CES.
- Double Barreled Amplifier (p. 36): Marshall Leach analyzes his design for a 250 W/ch kit with signal-tracking supply voltages and ostensibly superb performance.
- Digital Techniques in Sound Reproduction (p. 54): An intro to the uses of binary coding.
- Equipment Profiles (p. 62): Lirpa 5-Kg tonearm/cartridge (the annual April-Fool's parody). Fisher 4029 cassette deck (works OK at 1-7/8 ips, very good at 3-3/4 ips, but play EQ rolled off). Alpine 7307 car stereo receiver/cassette unit and 3002 power amp (tuner medium-sensitive, system otherwise excellent with honest and conservative ratings). Luxman K-12 cassette deck (an expensive two-head design, superbly well constructed, unusual control designs, counter actually displays tape timings, performance generally fine). Technics CO1 mini FM-AM tuner (mediocre sensitivity and subcarrier filtering, performance otherwise excellent).

Audio Amateur, 1980 No. 2

- The Grounded Ear (p. 2): On some of the more interesting new goodies seen at Winter CES.
- Power Supply Regulator for Op Amp Circuits (p. 8): Reasonably simple, yields very low supply impedance, important for the cleanest sound.
- Timerless Tone-Burst Generator (p. 14): CMOS logic yields flexible operation.
- Audio Windows (p. 19): A clear discussion of designing and making infrasonic, ultrasonic, and audio bandpass filters with selectable cutoff frequencies.
- Intensity Stereo (p. 29): Stan Lipshitz argues that Blumlein miking is the "only theoretically legitimate stereophonic recording system."
- Audio Aids (p. 34): Phono capacitance switching, de-humming turntables, etc.
- Test Report (p. 38): Hafler DH-200 power amp (measures very clean, 250 W at 4 ohms, up to 13 amperes of output current, very fast; Jung hears a slight HF sizzle using Magnepans with Fulton wires but not with conventional speakers and wires; sound is judged "powerful, detailed, dynamic, and musical"). SouthWest Technical Products 210A Tigersaurus power amp (hum, buzz, parasitic oscillation, two tested samples unlike each other).
- Circuit Schematics (p. 48): Audio Research SP-6A preamp, Dyna FM-5 tuner front end.
- Letters (p. 50): Improving the Advent MPR-1, et al.

The Audio Critic, Vol. 2 No. 2

- Letters (p. 2): Including a hilarious diatribe from Frank Van Alstine describing the flaws of his FET-5 Mk 5 preamp mod.
- Seminar, Part 2 (p. 9): A long-winded conversation about ultrasonic filtering, rise times, hypothetical time-modulation distortion in preamps, preamp bypass testing, etc.
- Wires and Cables (p. 23): A common-sense debunking job, with notes on some circumstances where low-inductance or RFI-shielded wire can make a real difference.
- Speaker Reviews (p. 28): Axiom TLT-1 (edgy upper-midrange because of wrong crossover design). Beveridge System 3 (low end colored by severe cabinet panel resonances, electro-static mid-top has wonderful imaging but a serious high-end rolloff, impedance falls to 1 ohm at 20 kHz; later samples may be better). B&W DM7 (spacious, transparent, largely uncolored, its only important fault is its high U.S. price). DCM Time Window (latest improved version is a clear best-buy at its price; transparent, smooth, detailed, can play *very* loud). Fried model C

(latest version is substantially improved; slightly edgy but rather good overall). Magneplanar MG-1 (no deep bass or extreme treble, severe ringing, but imaging is good). Onkyo F-5000 (smooth frequency response, but poor imaging, serious ringing, no time-alignment). Perspective Mk2 (measures flat, sounds awful, severe ringing, expensive, a rip-off). QLN 1 (transparent and focused, but crossover frequency is too low and impedance load is difficult to drive). Sound-Lab R-1 electrostatic (needs work, but a contender for "best," extremely transparent, uncolored, but hard for some amplifiers to drive). Swallow CM-70 (edgy and fatiguing). Vandersteen IIA (improved version the "best-focused, least colored, most balanced-sounding" dynamic speaker, but won't play loud).

- A Genuine Breakthrough (p. 35): A rave review of the $200 NAD 3020 integrated amplifier: "wipes out most $1000 preamps plugged into most $1000 power amps."
- Amplifier Reviews (p. 37): Amber 70 (best in its class, sounds a little better than the Hafler and PS). Audionics BA-150 (sweet, smooth, edgeless, but somewhat veiled; lots of IM measured). Audire Crescendo (transparent mid, edgy top). Audire DM700 (ditto). Bedini 25/25 (class A, sounds fantastically good under the right conditions). Bedini 45/45 (good but not great). Hafler DH-200 (followup report: production samples have varied, units after #3946000 are better). JVC 7050 (class super-A, sounds great, the best big amp). PS Model One (extremely good, competitive with Amber and Hafler). Sonotron PA-2000 (measures well, sounds spiky and fatiguing).
- MC Step-up Devices (p. 43): RWR MCT-1 transformer (almost as good as the Cotter), Marcof PPA-1 (the best head amp, a clear best-buy at its price, but not equal to the best transformers). Denon HA-1000 head amp (not recommended). Denon AU-340 and Signet Mk12T transformers (good but not great).
- Tape Deck (p. 45): Tandberg TD 20A open-reel deck (splendid in both mechanical performance and sound).
- Record Playing Devices (p. 46): DB Systems DBP-10 phono alignment protractor (neat, convenient, correctly calibrated, highly recommended). Dennesen Soundtraktor alignment protractor (ditto, but depends critically on your knowing the exact lateral pivot location). Denon 401 tone arm (OK, nicely made). Denon 303 cartridge (not as good as the 103D). Denon DP-80 turntable (it's fine when mounted in the Cotter B-i base). Discwasher Discfoot vibration isolators (worth a closer look). Fidelity Research FR-14 (gorgeously made, but no VTA adjustment). JVC MC-1 m.c. cartridge (smooth, transparent). JVC 7045 tone arm (well designed, VTA adjustable during play, may be a best-buy). Kenwood KD-650 turntable (improved version of the KD-500, relatively good arm included, isolation poor, becomes a best-buy system when mounted on a Cotter B-2 base). Linn-Sondek LP 12 turntable (overpriced, subject to low-bass feedback under the wrong conditions, parts of it are acoustically live). Ortofon MC-30 m.c. pickup (a big disappointment). Thorens TD 115 turntable (another disappointment). Wheaton 240 straight-line arm (nicely made, arm is hollow and resonant).
- FM Tuners (p. 51): Sequerra Model 1 judged best, with Yamaha CT-7000, Pioneer Series 20 Model F-27, and NAD 4080 close behind in audible comparisons, but FM broadcasting judged too poor to justify any serious tuner reviewing.
- Preamplifiers (p. 53): Audionics RS-1 (not first class but one of the best at its price). Audire Legato (pretty good but for an equalization error). Baumann Pro 4000 (measures terrific, sounds good but not great). Cotter System Two (costs $3280, not yet in real production, but is the reference standard). Hegeman HAPI Two (simply beautiful, second only to the Cotter and much more affordable). Precision Fidelity C7 (a tube phono stage plus volume control, extremely good, smooth, mellifluous). PS Audio III phono stage and LCC control center (transparent and neutral, an unequivocal best-buy, except for a mediocre m.c. head amp).

Audio Engineering Society Journal, April 1980

- Real Ear Tests for Stereophones (p. 206): Jon Sank describes his headphone test technique, incorporating psychoacoustic factors. Looks good.
- Low Noise M.C. Preamp (p. 219): Design of a head amp with very low noise using a special transistor with huge junction area.
- Digital Committee Report (p. 259): Includes a reprint of the EIAJ standard for digital converters used with VCRs.

Gramophone (England), April 1980

- Cassette Feature (p. 1511): Notes on recent developments in cassette technology, a survey of current pre-recorded tapes, details of Philips high-speed duplication facilities, and a review of the B&O 8000 microprocessor cassette deck (good except for an unexplained treble rise).
- Commentary (p. 1602): An interesting note reporting that with various digital recording systems the digital tape appears to be drier (lacking in high-frequency ambience) when compared to analog tapes.
- Report from America (p. 1607): Products seen at the Winter CES.
- Reviews (p. 1608): Dual 626 turntable with Ortofon ULM 55E cartridge (excellent overall, arm resonance at 12 Hz and well damped, cartridge tracks well and sounds pretty good but needs added capacitance for flattest response). JVC Zero 5 loudspeaker (generally good, with some upper-mid hardness).

High Fidelity, April 1980

- Editorial (p. 6): Against close-miking.
- Equipment Reports Ip. 18): Vector Research 9000 receiver (lots of buttons and controls, digital tuner works well, phono preamp OK except for nonstandard impedance, power amp has good headroom and current output). Dynavector 20A Type 2 high-output m.c. pickup (lightweight, stylus damping effectively controls arm resonance, internal capacitor damps ultrasonic resonance; response is smooth, detailed, balanced). Dynaco A250 speaker (fairly smooth, unusually uniform polar pattern, good value overall). Onkyo 2080 cassette deck (fine performance, auto-bias system works superbly, Dolby tracking excellent, mediocre metering). Koss HV/X headphones (exceptionally comfortable, excellent sound). ADC Sound Shaper Three equalizer (a graphic equalizer with unusual control flexibility, works fine).
- Into the Eighties (p. 43): Surveying the new products and prototypes shown at the Winter CES.
- Crises (p. 53): Traumas in the classical divisions of the record companies: background and prognosis.

Hi-Fi News & Record Review (England), March 1980

- Horns in the Home (p. 51): On making sub-floor horn subwoofers, plus notes on the horn speakers of a Sony executive.
- Equipment Review (p. 115): A comparison of three systems of "micro components" by Technics, Uher, and Sony. In power output, Technics CO1 measures best at 8 ohms, worst with 4 ohms or a reactive load; Uher Z140 best with difficult loads. Preamps OK except Uher VG-840 has poor phono overload and S/N. Sony P7J the best tuner in group except for faint audible tones leaking from digital synthesizing circuits.
- Two high-end cassette decks: Tandberg TCD 440A (ergonomics unusual and well-liked, DYNEQ high-frequency limiter works splendidly to prevent saturation; very good S/N, superb sound especially with metal tape). Pioneer CTF 1250 (flexible bias/EQ adjustments, some tendency toward under-biasing, less headroom than the Tandberg, response generally very flat, overall sound quality excellent).

HiFi Stereophonie (Germany), March 1980

- Opera Choirs-An Mass Ornament or People's Voice? (p. 284): About the changing role of the opera choir.
- The Anonymous Opera Protagonists (p. 289): An analysis of the economic situation of the opera choirs in Germany.
- Musical Actors or Singing Walker-Ons? (p. 290): The opera business in view of the choir at the Vienna State Opera.
- Benjamino Gigli, the Rat-Catcher of Recanati (p. 294): Critical remarks to the complete discography, just released by EMI Italiana.
- Disco as a Ritualized Free Time (p. 301).
- Small Labels: FMP, Enja, SteepleChase (p. 308).
- Test Reports (p. 358): Three mini components, the Hitachi ACT-M2, the Korting Series 100, and the Siemens Hi-Fi-System 666 (good at their respective prices), the Luxman K-12 cassette recorder (weak in several respects), the Nakamichi T-100 Audio Analyzer (very fine and versatile), The Akai SR-1100, the Acoustic Research AR25, the B&O Beovox S45-2, the Hitachi HS-3, the Sharp Optonica CP-2711, and the Technics SB-F3 loudspeakers in a psychometric test (the Sharp Optonica ranked first).
- Shrill Sound, Annoying Hum (p. 403): Things not to be overlooked in setting up the phono equipment, namely the interface aspects.

HiFi Stereophonie (Germany), May 1980

- Inflation of Music Festivals (p. 560): The aspects of this manifold phenomenon discussed.
- Records and Music Festivals (p. 562).
- An Interview with the Intendant (General Manager) of the Berlin Festival (p. 564).
- Folk and Minstrel Festivals Between Consumption and Contact (p. 568).
- Bombay Jazz-Yatra '80 Festival (p. 571).
- Winter Music '80 in Baden-Baden (p. 576).
- Cassette Recorder Anatomy (p. 611): The basics of its typology, electronics, transport mechanism, and recording level indicators explained.
- Test Reports (p. 632): Six cassette recorders compared: the Akai GXC-706D (good), the Dual C 810 (second best in this group), the Grundig CF 5000 (good, but the operating flexibility is a bit limited), the Philips N 5331 (meters rather slow), the Sony TC-K 35 (best of the group), and the Uher CG 310 (good).
- The 65th AES Convention in London (p. 653).
- Festival du Son 1980 in Paris (p. 658).

Modern Recording, April 1980

- Reviews (p. 70): Sansui B-1 power amp (expensive, powerful, clean, conservatively rated). ADC Sound Shaper Three graphic equalizer (flexible tuning of band frequencies, works fine, measures well). Nakamichi 680-ZX cassette deck (superb at standard spped, amazing at half-speed, obviously the best deck on the market especially with metal tape; low-end extends to 10 Hz). Ursa Major Space Station digital delay/reverberation unit (better than most other electronic reverberators, not as good as the best spring reverbs).

Popular Electronics, April 1980

- Stereo Scene (p. 14): Real and mythical differences among amplifiers, plus notes on recent super-discs.

- Test Reports (p. 22): Dual 606 turntable with Ortofon ULM 55E pickup (superb tracking of normally unplayable warped records, system resonance around 12 Hz but quite broad, good arm geometry, effective mass with cartridge only 7.5 grams, cartridge is pretty good, needs extra capacitance for flattest response). Mitsubishi MS-40 loudspeaker (neutral, uncolored, measures somewhat bright). Audio-Technica 1010 tone arm (medium mass, good geometry, resonance damper, outstanding tracking of warped discs).
- Sound and Specs (p. 38): A thought-provoking review by Daniel Queen of why it is very difficult to develop specs which correlate well with a loudspeaker's real-world subjective performance, and another by Bob Berkovitz noting some of the difficulties of reliable subjective evaluation and reporting on a remarkable experiment (cross-correlating signals from mikes in a dummy head) which mimics our perception of localization and maps it objectively.
- 3-Way Drive System (p. 46): A homebrew electronic crossover.

<u>Stereo Review, April 1980</u>

- Audio/Video News (p. 26): David Ranada reports on the first dbx encoded discs from digital masters: wonderful potential, not fully realized in the initial issues.
- Audio Basics (p. 34): A <u>really</u> basic intro to preamps.
- Tape Talk (p. 36): Answers about dropouts, fugitive highs, etc.
- Technical Talk (p. 41): Debates about stereo imaging.
- Test Reports (p. 42): KLH 1 speaker (swaybacked response curve with overwhelmingly strong bass, can be played extremely loud without distress, analog "computer" optimizes bass output while protecting woofers against overdrive). Sonus Dimension 5 phono cartridge (tracks well at 1.5 g, top-end response rises to an undamped 30-kHz peak, sounds like a moving coil). Philips AH180 digital tuner (very good performance, extremely flexible digital tuning, will not be obsoleted). Spectro Acoustics 200SR power amp (very conservatively rated, has lots of dynamic headroom and lots of peak output current, no current-limiting, a best-buy candidate except that its IHF slew factor is only 2.0, so it should be used with an ultrasonic filter in the preamp). Eumig FL-1000 cassette deck (expensive, superb, auto-bias system works well, can be interfaced with computers for extreme operating flexibility, flutter is ultra-low).
- Equalizers (p. 72): Dan Shanefield discusses the assorted uses of EQ.

- Peter Mitchell (Massachusetts) and Jiri Burdych (Czechoslovakia)

March BAS Meeting

Business Meeting

The March 23, 1980 general meeting of the BAS was held at GTE Labs in Waltham. Society President Peter Mitchell began the business session with a summary of the most recent Executive Committee meeting (March 15, postponed from February 28) decisions: the BAS has accepted a subscription to Wireless World; the Ivie spectrum analyzer has been ordered; the Test Equipment Committee will henceforth be a permanent standing committee-this means that chairman John Schlafer is now a member ex officio of the Executive Committee; Peter mentioned that the new draft BAS Constitution and Bylaws are to be presented to the membership at the April general meeting. The next Executive Committee meeting was scheduled for April 13.

The Program Committee has developed a suggestion form listing among other items possible candidates for speakers at future meetings, subjects for meetings, test clinics, and other BAS activities. Your participation is requested.

On the subject of publicity, it was announced that the BAS will run paid advertisements in the Boston Phoenix and the Real Paper (local "alternative" tabloid weeklies which often run hi-fi supplements). Also, a BAS promotional flyer is now ready from the printer. These flyers will be distributed in local and (hopefully) outlying audio stores, etc., with the goal of inducing more audiophiles to join the Society. Members (especially those from the hinterlands) are urged to help distribute the flyers. Quantities may be picked up at future BAS meetings.

Jack Stevens is exploring ways to computerize the BAS mailing lists in the face of anticipated membership growth. He is interested in a good, reasonably cheap microcomputer system (used, perhaps?). In addition to maintaining the mailing lists and reducing the drudgery for Henry Belot, Frank Farlow, Jack, and others, the computer system might be used in conducting experiments on audio equipment (e.g., analyzing loudspeaker impulse response measurements, etc.). Anyone with helpful suggestions should contact Jack Stevens.

Open Forum

Al Foster began the Open Forum session with an offer of $5-per-foot "Mogami" speaker cable to someone willing to test the stuff and report on its effectiveness in a hi-fi system. Al went on to remind members that last November the nice people from Empire Scientific distributed various "goodies" on the understanding that recipients would report back their results of use. He complained that no such reports had been received yet.

Scott Kent announced that the list price of his records is going up-to $9; therefore, his discount prices to meeting attendees will go up from the current $ 5.

Peter Storkerson reported that WBUR has improved its facilities again: the station now uses SME Series III tonearms and has installed a new stereo exciter.

Louis C. Souther, d/b/a Southern Engineering Products, displayed a pair of homebrew tone-arms: first, an operational wooden version of the Dynavector DV-505 arm (the huge, complicated thing)-Mr. Souther claims considerably improved tracking ability for his ultra-lightweight model; second, a not-quite-completed straight-line-tracking arm system of his own design, which he promised would be ready for demonstration in the near future. Interested members might get in touch with Mr. Souther at upcoming meetings.

Finally, several members discussed some of their experiences with video equipment, e.g., difficulties in duplicating videocassettes, incompatibilities between different video recorders, problems encountered in receiving certain TV channels, etc. Those seeking further details are encouraged to attend future BAS meetings.

Meeting Feature

The featured speaker was Stephen F. Temmer, founder and President of Gotham Audio Corporation, New York. Gotham Audio is the U.S. distributor for several lines of German professional recording equipment: Neumann, EMT, Telefunken, and DIN test records, among others. Gotham Audio is 22 years old, but most of us didn't become aware of Stephen Temmer until about a year or so ago. That was when he effectively stalled the Audio Engineering Society Digital Standards Committee by wondering out loud (and within earshot of the Justice Department) whether that committee's activities might not violate antitrust laws. [See the February 1979 Speaker (page 5) and also Mr. Temmer's comments in the May 1979 Speaker (page 4).] The resulting flap earned him, among other things, an invitation to address the BAS.

Mr. Temmer, born in Austria, was once a Vienna Choir Boy. He studied piano with Moriz Rosenthal (a pupil of Franz Liszt) and plays a Bosendorfer piano at home. His education includes a short stint at M.I.T., but he doesn't consider himself a technician. Rather, he relies on his musical ear to tell him whether things sound good, although he admits that the older he gets, the less inclined he becomes to pronounce judgment about what things sound like. His standard answer to hi-fi demonstrations is, "It's not obviously bad."

It is fair to say that Mr. Temmer is still a sufficiently opinionated person to infuriate some people in the audio industry, especially the manufacturers of professional digital recording equipment. In Stephen Temmer's perfectionist view, the people in the pro audio industry have an obligation not to compromise their standards for consumer purposes. A year ago he sensed a "clear and present danger" that some in the industry, eager to promulgate the use of digital technology both in the interest of better sound and for commercial gain, would push through a half-baked digital standard which would impede further progress.

His major complaint is against the proposed sampling rate standard, which would be somewhere in the vicinity of 50 kHz. Now it happens that for good technical and economic reasons, a 50-kHz sampling rate is just about the best that pro audio manufacturers can build and sell these days. That rate is compatible with current and proposed digital telephone and satellite communications systems. It is convenient, but according to Temmer is not based on listening criteria. In his opinion the rate should be at least 100 kHz, and maybe as high as 200 kHz. At a sampling of merely 50 kHz or so, he finds that the high audio frequencies "grate on my nerves"--above 2-3 kHz the sound is "discontinuous." Although most people are impressed by today's digital sound, he predicts we will all be tired of it in another 5 years. Mr. Temmer has heard experiments which demonstrate to him that as the sampling rate goes up, the sound becomes smoother, "less harsh."

The ear is educable. (Anyone who has learned to play a musical instrument, or has repeatedly upgraded a hi-fi system, knows what that means.) Mr. Temmer gave an instructive example. Some years ago he (and others, including M.I.T. Professor Francis F. Lee) helped found the company now known as Lexicon. They built the first commercial digital delay line system. The first model had 60 dB of dynamic range; even with a great deal of pre-emphasis, it sounded awful. Adding more bits to get 72 dB made little improvement. Even at 96 dB it isn't perfect and still needs pre-emphasis. (Note: in a PCM system, each additional bit per sample adds about 6 dB of dynamic range.) The point of this story is that the customers thought the system impressive at first; a year later they were unhappy with the sound and clamoring for a better "next generation."

In short, Mr. Temmer doesn't think we truly understand digital technology well enough to agree on standards yet. Standards impede progress. He likes the videocassette market, which is doing very well in spite of six competing systems. Such a situation may be economically expensive, but it does lead to rapid technological progress. He noted the historical consequences of too-early standardization: the English railway system (narrow-gauge, incompatible with the rest of Europe) or the United States NTSC color television standard (the Europeans use the much better PAL standard, developed after it became clear what was wrong with NTSC). He thinks we would do well to drag our feet on the digital recording standards issue-let the Japanese devise their own premature standard if they must; we should wait until we know what we're doing. As chance would have it, the AES Digital Standards Committee is now bogged down in minutiae. Peter Mitchell noted that they appear to be relieved they no longer need to rush to judgment, so it appears that Stephen Temmer can relax.

(Editor's note-According to Mitchell, there are about eight different standards up for consideration by the Committee, each with plausible arguments in its favor. Much of the argument concerns not whether the sampling frequency should be 50 or 100 kHz, but exactly where in the neighborhood of 50 kHz it should be. What is sought is a sampling rate and accompanying word length which will divide evenly into various time intervals used for the NTSC video frame rate, the SMPTE time code, the frame rate of film, the 32-kHz sampling rate used in the BBC's relay system, the European telecommunications system, and so on. The task is immense and promises to take years.

Meanwhile the Japanese, who have been working longer and harder on this problem than the Americans, have settled their internal differences and can present the AES Committee with a complete standard, including sampling rate, word length, data format, and error correction methods.)

Digital Disc Technology

Before reporting on Mr. Temmer's second topic for the evening, the Teldec digital disc, it seems desirable to provide a little background information on the subject of digitally encoded discs (both video and audio, since the technology will accommodate either type of signal). There are several incompatible systems available today; they differ significantly in the manner of storing the digital data on the disc surface and in the method of playback. Three basic kinds of playback systems exist: optical, capacitive, and mechanical (piezoelectric).

The most common optical playback system is that developed by Philips and MCA, now available in the MCA "DiscoVision" system. Here the information is stored in the form of reflective pits on a reflective background. The data is scanned by a focused laser beam. The presence or absence of a pit modulates the reflected beam, which is sensed by a photodetector and yields the digital signal after appropriate processing. Such a system is wear-free (as far as the disc is concerned). Its major disadvantage lies in its technical complexity, which must translate into high cost. The playback system requires carefully aligned laser optics. A "DiscoVision" disc surface is actually a three-layer sandwich: a pitted substrate, a thin reflective metallic layer, and a transparent protective layer. An advantage of this construction is that dirt, fingerprints, etc., on the outside of the protective layer are out of the optics' focal plane and will not degrade playback. Philips is about to introduce the technology in its "Compact Disc" audio-only system.

The capacitive playback system can be made more cheaply. In this scheme, a stylus rides on the information-bearing surface. In the RCA system, the surface is grooved, not unlike a conventional phonograph record; the JVC system uses no grooves. The stylus acts as one plate of a capacitor, the disc itself as the other; this implies that the disc surface must be an electrical conductor. The information is recorded vertically into the surface. As the stylus rides over the undulations, the capacitance of the stylus/disc system is modulated; these capacitance variations can be detected to recover the recorded signal. A drawback with this scheme is that the disc surface must be kept absolutely clean; these discs come housed in a protective "caddy."

The mechanical playback system is the simplest and cheapest of all. It uses the principle of the piezoelectric phonograph cartridge; only the recovered information is in digital form rather than analog. The disc material need have no special electrical or optical properties-high quality PVC will do. Again, it is necessary to protect the record surface from contamination.

There is widespread acknowledgement that Japanese industry is besting its U.S. counterpart in terms of productivity growth, innovation, research and development, marketing, quality control-you name it. The results are obvious when you walk into any consumer-electronics store. Stephen Temmer would agree that U.S. manufacturers are not doing a good job of competing in the consumer market. He cited the German hi-fi market as a good example of what ought to be done instead. There, he says, the German manufacturers have challenged the Japanese and are winning; he offered the Teldec digital disc as an illustration of what is possible.

Teldec apparently began working on a videodisc system in March 1966, after hearing a false rumor that CBS was starting to work on the same problem. By the early-mid 70's, Teldec had a product; it was a technical triumph and, alas, a marketing flop. Today it still survives as an audio-visual instructional device for German doctors. Teldec has now adapted the technology to an audio PCM disc. Some of the specifications for the playback system are as follows:

Disc material	High-grade PVC
Playback system	Piezoelectric
Stylus pressure	1 milligram
Disc diameter	13.5 cm (5.3 in.)
Playing time	60 min/side (both sides playable)
Audio channels	4
Groove spacing	600 grooves/mm (15,240 grooves/in.) 1.66 micron/groove (0.065 mil/groove)
Groove speed	1.89 m/sec
Rotational speed	Between 278 rpm (near edge) and 695 rpm (near center)
Resolution	0.61 micron (0.024 mil)
Storage density	Approx. 1 Megabit/mm^2 (645 Megabit/in.^{2})
Transmission bandwidth	3.072 Megabit/sec
Sampling rate	48 kHz
Quantization	14-bit linear
Dynamic range	85 dB
Channel separation	120 dB
Redundancy	8%

The master is cut into a copper blank. No "horns" result, as when cutting into lacquer. The grooves are vertically modulated by the digital information. The "chip" that results from this cutting process is finer than any copper wire that can be drawn, and is salvaged and sold by Teldec. While the master is centered on the lathe, another tool is used to introduce a locating shoulder (referenced to the same center as the grooves, of course) in the center of the disc. This ensures that each side of the final product can be perfectly centered in the playback system.

(A great idea. Why hasn't anyone tried that before?) This master is used as a "mother" to make any number of stampers, which are then used to press the vinyl end-products in more or less conventional fashion. The disc is held in place in the playback unit by a ceramic magnet which attracts a metal ring pressed into the center of the disc.

The playback mechanism consists of a remarkably simple electromechanical transducer. A small sled-shaped chip of piezoelectric crystal rides (or skates) over the vertical modulation of the disc surface. A shock wave is generated at the trailing edge of the crystal as the vinyl rebounds from the pressure of the pickup. This produces several millivolts of output from the crystal, from which the recorded digital information can be recovered. The total stylus force is about 1 milligram, so that record wear is just about negligible. The piezoelectric crystal will eventually dull; therefore, a sharpening groove lined with diamond dust is built into the playback unit. A few times around and the stylus is good as new. Currently, the discs are being pressed on an RCA vinyl compound which contains carbon black, and is therefore conductive. So (in principle, at least) the discs could be played back on systems using either mechanical, capacitive, or optical pickup systems. Mr. Temmer claims that the Teldec scheme provides the best price/performance ratio available; it is ingenious and simple at the same time. By comparison, other systems appear over-engineered and will certainly cost more to repair.

Having available four discrete audio channels opens up all sorts of possibilities: discrete four-channel sound, separate ambience channels, two ordinary channels and two binaural channels, separate channels for soloists or vocalists, etc., all with total channel separation.

So what's wrong with it? Well, for one thing, Mr. Temmer dislikes the 48-kHz sampling rate. For another, it doesn't really exist. Telefunken apparently has no plans to market the system in the U.S.; indeed, the company seems to be facing hard times in Europe, so they may have marketing problems there, too. U.S. audiophiles may eventually have to settle for the nice-but-expensive Philips laser "Compact Disc," or for something from RCA, which appears to have no specific plans for an audio-alone system. Meanwhile, hang on to your analog discs. Stephen Temmer predicts they'll be around for a long while yet.

Miscellany

Toward the end of the evening, Mr. Temmer delivered himeslf of opinions on a variety of subjects of interest to him and to B.A.S. members:

1. Record Quality--Commercial records are what they are (poor to mediocre) because the public will buy them that way. "Good enough is what makes the most money." If records had to be better, they would cost more and fewer would be sold. Current price and quality levels yield the highest return on investment for the record industry. What would it take to improve the situation? "The only missing ingredient for quality is money." In Japan, consumers will pay $20 per record; in return, they demand the highest quality. And they get it.

2. Noise Reduction-Its use is an admission that the end of the road has been reached, technically, in analog recorders. In commercial recordings, most of the dB's available from noise reduction have been spent in (cost-cutting) degradations elsewhere. The same results could have been achieved with extra care (i.e., money), which, however, no one wants to spend in the commercial world.

3. Transformerless microphones-A transformer is the most effective shield against RFI, of which there is much in commercial recording situations. RFI generally degrades signal quality and S/N ratio. However, transformerless microphones definitely sound better. Neumann is in the process of removing input (low level) transformers from its lines of microphones. Output transformers will stay, however.

Mr. Temmer concluded his interesting presentation with playback of a copy (third generation) of one of the first true stereophonic recordings made: a performance of a Beethoven piano concerto (No. 5) by Walter Gieseking et al. made in October 1944 in Berlin using 3 Neumann CMV3 "bottle" microphones (a 1928 design still in use in places today) and an RE2 tape recorder. Aside from poor bass, especially considering the date and circumstances of the recording; anti-aircraft guns could be heard in the background noise. (Note: this recording has been released on LP recently. It is reviewed in the January 1980 Hifi News & Record Review on page 119.)

- Pieter Bras

Acoustical Comb Filter Effects

F. Alton Everest

The following article is reprinted from Recording engineer/producer magazine, Volume 10, Number 4 (August 1979). Re-e/p, as it is known in the trade, is sent free of charge to qualified recipients in the U.S. Those not connected with the pro audio business may subscribe by sending $10 for surface mail delivery, $17 for air mail, for one year (six issues) to: Gallay Communications, Inc., 1850 N. Whitley Avenue, Hollywood, CA 90028.

This particular issue contains a series of articles on reverberation, both natural and artificial, by BAS meeting guest Alan Fierstein, local member David Griesinger, and others.

F. Alton Everest is a well-known acoustical consultant and author of many reference books on professional audio; one of his best-known works, Acoustic Techniques for Home and Studio (Tab Books, Summit, PA, 1973) is an excellent survey of sound in small and large rooms, with many practical suggestions for construction and evaluation.

The author has advised us that the following corrections should be made to the piece as printed:

1. The caption of Figure 1 should read: "The diaphragm of the microphone responds to the vector sum of sound pressures from A and B."
2. Figure 2(A) is for 0.1 ms delay and Figure 2(C) for 1.0 ms delay.
3. Figures 9(C) and (D) are reversed.

We would like to thank the publishers of Recording engineer/producer and Mr. Everest for their permission to reprint this article.

E.B.M.

emphasis: REVERBERATION

Acoustical COMB FILTER Effects

(... *how to keep them out of your hair ...*)

by

F. Alton Everest

Phasing and flanging are certainly well known today and those using such effects generally associate them with the term **"comb filter."** Less widely apprciated is the fact that in many common recording and reproduction situations comb filter effects mess up our desired flat frequency response. This is a form of amplitude distortion which is inherent in practically every listening and monitoring setup, every single or multiple microphone mono pickup, as well as many mixdowns from stereo to mono. The magnitude of this distortion depends primarily upon the geometry of the setup, although other factors enter in.

Delay is the key word!
sound as an acoustical phenomenon, delay is a direct result of the finite velocity of sound. For normal temperatures and near sea level sound travels about 1,130 feet per second, or 1.13 feet per millisecond. In evaluating practical problems, a very convenient thing to remember is that sound travels about one foot per millisecond.

A microphone is a rather blind sort of instrument. Its diaphragm responds to whatever fluctuations in air pressure occur at its surface. If the rate of such fluctuations (frequency) falls within its operating band it obliges with an output voltage proportional to the magnitude of the pressure involved. If a 100 Hz tone from a loudspeaker actuates the diaphragm of a microphone in free space, a 100 Hz voltage appears at the microphone terminals. If a second loudspeaker lays down a second 100 Hz signal at the diaphragm of the microphone identical in pressure, but 180° out of phase with the first signal, one cancels the other and the microphone voltage falls to zero. If an adjustment is made so that the two identical 100 Hz acoustical signals are in phase, the microphone delivers twice the output voltage, an increase of 6.02 dB. The microphone slavishly responds to resultant pressures acting on its diaphragm. Little did it know (excuse the anthropomorphism) that when the two identical 100 Hz acoustical signals were in phase opposition that air molecules a short distance away from the diaphragm were obediently doing their violent 100 Hz dance. In short, the microphone responds to the vector sum of air pressure fluctuations impinging upon it. We must remember this characteristic of the microphone as we dive into a considera-tion of acoustical comb filter effects.

-the author —

F. Alton Everest has been involved in sound and acoustics since the mid-thirties. He has been involved in the research of acoustic problems as well as the practical applications of their solutions. He has authored numerous books including the "Handbook of Multichannel Recording" (Tab), "Handbook of Public Address Sound Systems" (Tab), and "Acoustic Techniques for Home and Studio." His achievements over the years are numerous and include co-founding the Moody Institute of Science, Whittier, California, where he was director of Science and Production for twenty-five years.

A Description

Now, as an astounding revelation to those who are not sure just what a comb filter is, and as a review to those patient ones who do, we shall examine this ubiquitious effect in detail. We have seen that when two different airborne acoustical waves, A and B of Figure 1, arrive at a given point in space, such as our microphone diaphragm, they combine vectorially, that is, with due regard to amplitude and phase. If A and B are identical sine waves of approximately the same amplitudes we have a highly simplified situation. With the 100 Hz example, combining in phase doubles the amplitude, combining in phase opposition results in cancellation. The same is also true in combining identical, but highly complex, signals.

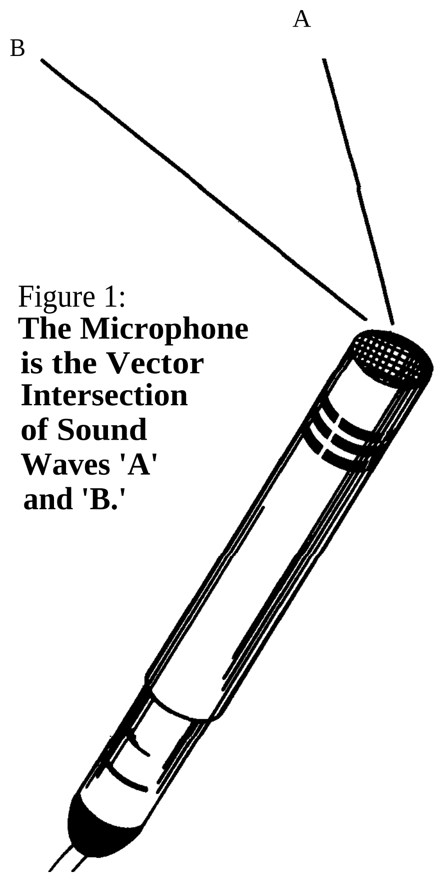


Figure 1: **The Microphone is the Vector Intersection of Sound Waves 'A' and 'B.'**

It is helpful to consider how this interference effect acts down through the audio spectrum. Comb filter interference can radically affect the overall frequency response even though the system compo-nents are flat. Let us assume that a microphone diaphragm is actuated by the combination of two signals, a signal direct from the mouth of one talking, and the same signal delayed 0.1 millisecond. Without the delayed signal let us say that the system response is flat and represented by the straight line at 0 dB in Figure 2(A). Adding to a given signal the same signal delayed 0.1 ms, the response undergoes some surprising changes. At those frequencies at which constructive interference takes place the response is boosted 6 dB. Midway between the 6 dB peaks, destructive interference creates dips infinitely deep, theoretically, 20 or 30 dB deep in practical situations. Significant energy is removed from the

Figure 2a

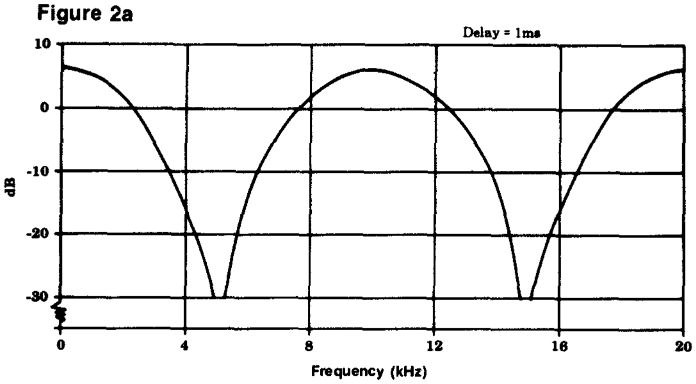


Figure 2b

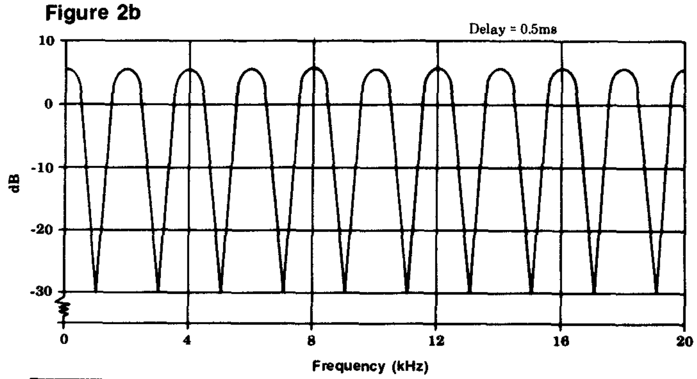
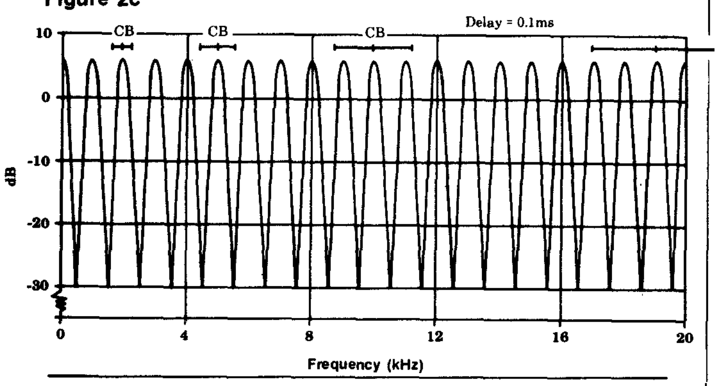
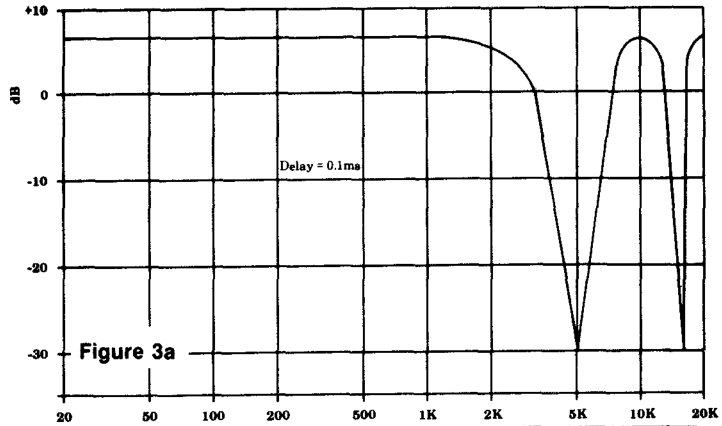
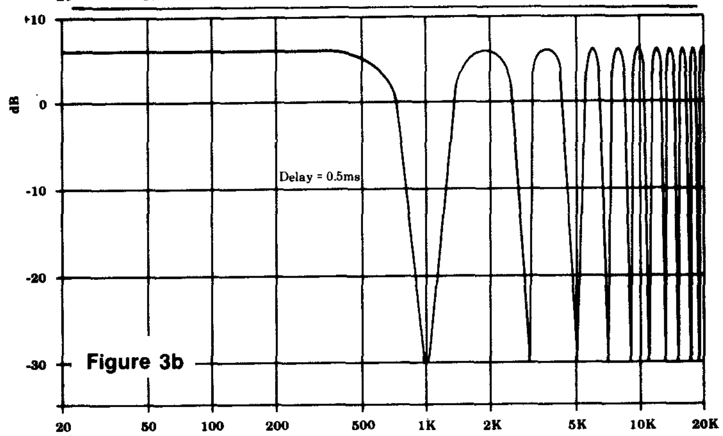


Figure 2c







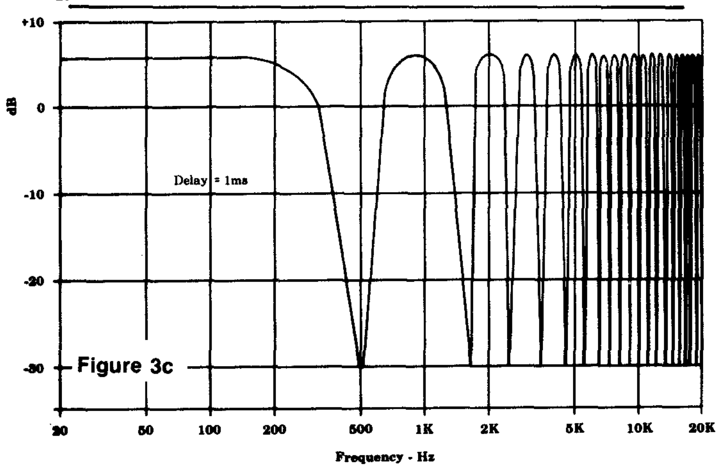
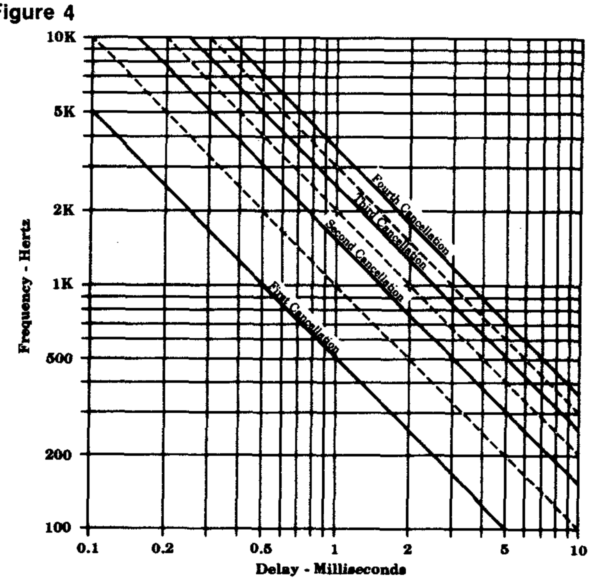


Figure 4



signal spectrum in the vicinity of 5 kHz and 15 kHz and unnatural peaks introduced below 2 kHz, above 18 kHz, and in the 9 - 12 kHz region. Note that a linear frequency scale is used to show the symmetry of the peaks and dips.

If the delay is increased to 0.5 ms the peaks and dips are much closer together as shown in Figure 2(B). It is now apparent why the comb filter name was applied to this effect. Peaks now occur at 2, 4, 6, and 8 kHz and at every other 2 kHz interval up through the spectrum. Between each pair of peaks is the accompanying dip.

Figure 2(C) illustrates the comb filter effect when a signal is combined with itself delayed 1 ms. Peaks are now separated 1 kHz, as are the dips.

Now that it has served its purpose of showing the inherent symmetry of the comb filter response, let us abandon the linear frequency scale for the more familiar logarithmic scale. Figure 3(A) shows the 0.1 ms delay case plotted in conventional semilog form. This gives a much better "feel" as to the effect of 0.1 ms delay on signal quality. The notches at 5 and 15 kHz would significantly color both speech and music. The 6 dB increase in level as a widening of the effective width of the dips.

The 0.5 ms case is presented in Figure 3(B) on semilog coordinates. The dips appear very narrow in this plot, especially at the higher frequencies. Readers who have had experience with controlling feedback frequencies in sound reinforcement systems by applying numberous narrow notch filters might say that the dips of Figure 3(B) might be tolerable, but not welcome. No matter how it is viewed, it is a significant deviation from a flat response.

Figure 3(C) illustrates the 1 ms delay example on a lot frequency scale. Dips at 500, 1,500, 2,500 Hz, etc., are interspersed with peaks at 1, 2, 3 kHz, etc. Looking at all three parts of Figure 3 we note the general principle that the longer the delay, the more the dips extend toward the low frequencies. Table I tabulates the location of the first null and spacings between adjacent nulls and adjacent peaks for delays from 0.1 ms to 50 ms. The same information in graphical form

TABLE 1
COMB FILTER PEAKS AND NULLS

Delay Ins	*Frequency of Lowest Null Hz*	*Spacing Between Nulls Spacing Between Peaks Hz*
0.1	*5,000*	*10,000*
0.5	*1,000*	*2,000*
1.	*500*	*1,000*
5.	*100*	*200*
10.	*50*	*100*
50.	*10*	*20*

is shown in Figure 4. The broken lines between the cancellation lines of Figure 4 show, of course, the location of the peaks.

Effect of Relative Amplitudes

In Figures 1, 2, and 3 it has been assumed that both the direct and the delayed signals are of equal amplitudes. In practical situations in which the delayed signal is the result of a less than perfect reflection, or arrives at the microphone at an angle at which response is down (e.g., off axis on cardioid mike), the delayed signal may be reduced in amplitude. Further, the inverse square law hasn't been repealed, in fact it hasn't even been called out of committee. A component travelling farther arrives at lower amplitude than the component of signal travelling a direct path.

The boost will be less than 6 dB above normal and the nulls will be less than minus infinity if the delayed signal is less than the direct. Considering these two amplitudes as a ratio of unity or less, Figure 5 enables one to determine theoretical peak height and null depth. If the delayed component is 80% of the direct, the peak height is above 5 dB and the null depth about 14 dB.

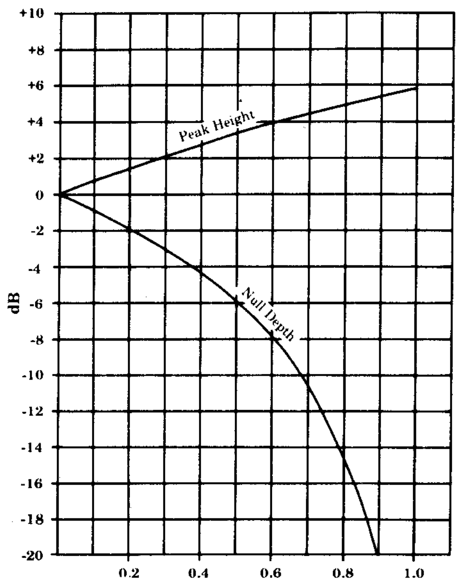


Figure 5

Comb Filters In Practice

There are many ways to generate comb filter effects. An easily manipulated method is that of Figure 6 in which a signal and a

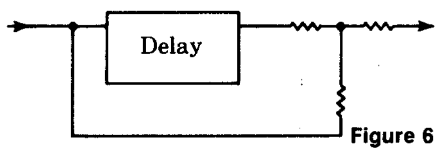


Figure 6

delayed version of itself are combined in a linear network. Applying a repetitive swept sine wave to the input and observing the output on a cathode ray oscilloscope for different delays, the responses of Figures 2 and 3 are readily reproduced.

Example #1

Getting out of the laboratory and into the real world, consider the podium microphone arrangement of Figure 7. Believe it or not, such arrangements with both mikes. feeding into the same amplifier can still be found.

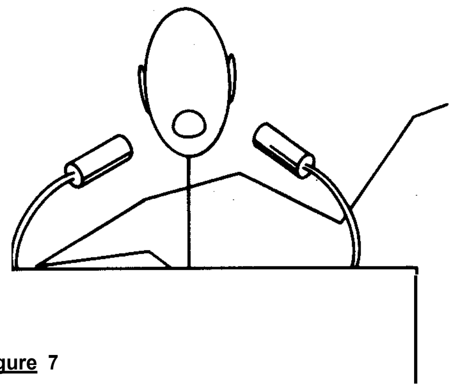


Figure 7

The excuse for using two mikes this way is that it gives the talker greater freedom of movement, but how about interference effects? Assuming the microphones are connected in phase (and they might not be if this practice is typical of the local personnel), and if the talker is dead center, there would be a helpful level boost of 6 dB. What happens if he is 3 inches off the center line? Let us assume further that the microphones are 24 inches apart and that the talking lips are 18 inches from a line drawn through the two microphones and on a level with the mikes. If the talker is centered, the sound travels the same distance (21.63") to each microphone. If the talker moves laterally 3", he comes closer to one microphone (20.12") and increases his distance to the other (23.43"). The difference between these two distances (3.31") results in the sound arriving at one microphone about 0.2 milliseconds behind the other. Result? A comb filter with a nice null gouging out important speech frequencies. If the talker were clamped in this position, the speech quality would not be good, but it would be unchanging. Normal talker movement results in very noticeable changes in quality as the nulls and peaks shift up and down the frequency scale.

Welcome evidence that such effects are real and not just theoretical scare tactics has come out of the Electro-Voice anechoic chamber. Lou Burroughs gives numerous examples of wildly distorted responses due to what he calls "acoustic phase cancellation," or comb filter effect, measured in simulated setups[2].

Example #2

Another situation much more common than Example #1 but similar in principle is

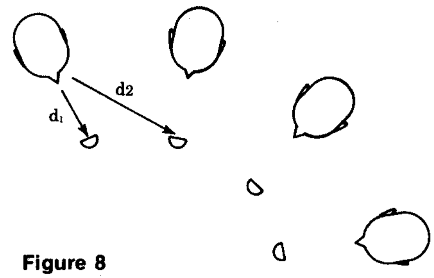


Figure 8

the case of multiple sources close together, each source associated with its own microphone. Let us consider the singing group of Figure 8 with each of the four singers having his or her individual microphone. Singer A is d1 inches from his or her own mike and d_2 inches from the next mike. The voice of A, picked up by both mikes, is mixed in the mixer with all the comb filter effects resulting from the path difference between d_1 and d_2. As each singer's voice is picked up, one degree or another, by all microphones, the situation gets more and more complex. Fortunately, sounds picked up by the more distant microphones are weaker and the comb filter boosts and dips are correspondingly reduced as per Figure 5. The experiments reported by Burroughs[2] indicate that if singer A's mouth is at least three times farther from the neighboring mike than from his own, the comb filter effects are negligible. If all singers place their microphones inside their mouths (some come close to this), the comb filter effects are submerged by other problems. If the "proximity effect" of the microphone boosts the bass, perhaps this amount of proximity will completely eliminate the highs as well as comb filter effects.

Example #3

Single microphones, like single men and women, can have their problems. Concentrating on the microphone side of the analogy, reflecting surfaces result in signals arriving at the microphone somewhat later than the direct signal. Figures 9(A) and 9(B) illustrate the case of the talker or singer standing before a microphone on a floor stand, or even hand held. In this case the floor reflected component (d_2) is much weaker than the direct (d_1) because (a) the distance of travel is greater, (b) the angle of arrival at the microphone is off the main axis, and (c) there is energy lost at the floor reflecting surface. Let us consider two simplified, specific cases, one in which d1 10" (Figure 9(A)), and another with d1 = 50", both having a microphone height and soloist mouth height equal to 56", and a floor reflection coefficient of 0.95. We shall also assume a cardioid microphone which would give a response at 90° about 3 dB less than on axis. With the source 10" from the microphone, the floor reflected component is delayed about 7.6 ms, but this information is significant only if the amplitudes are reasonably comparable. Considering only path length differences (10" direct vs. 112.4" reflected) we would expect the reflected component to be about 21 dB below the direct due to spherical divergence (inverse square law). The reflection loss at the floor is only about 0.4 dB and the 90° off axis cardioid loss is another 3 dB. This places the reflected component more than 24 dB below the direct. Obviously, the resulting comb filter interference effect would be negligible.

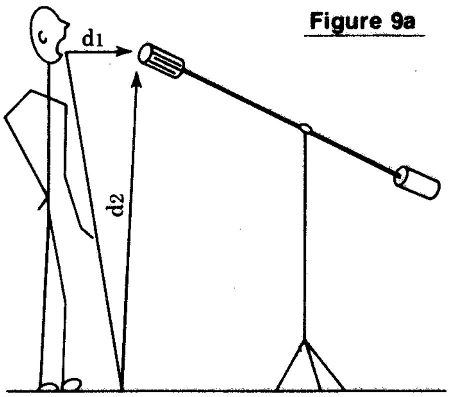


Figure 9a

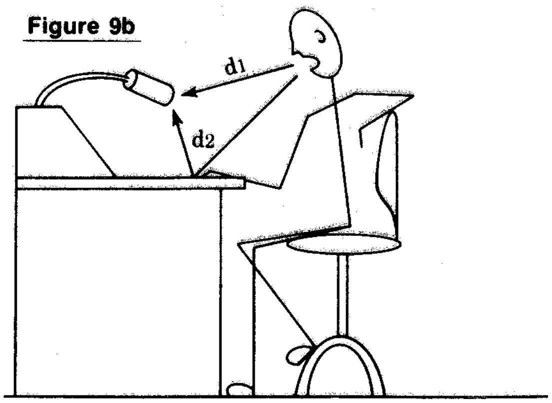


Figure 9b

The soloist now moves back to a point 50" from the microphone, either to improve "ambience," simply shrinking in fear, or for some other reason good enough for us to get on with this example. The direct path (50") is now somewhat more comparable to the reflected (122.7") and the reflected component arrives at the mike about 5.4 ms later than the direct. This would put the first null around 100 Hz, the second one near 300 Hz, which could be serious if the amplitudes are close enough in magnitude. It turns out that the inverse square loss is about 7.8 dB to which we must add 0.4 dB for floor reflection and something like 1 dB for cardioid pattern 66° off axis making a total of something like 9.2 dB. This corresponds to a ratio amplitude of about 0.35 which, from Figure 5, indicates we can expect nulls about 4 dB deep and peaks about 2 dB high, or overall perturbations of our response of about 6 dB.

There are things we can do to improve the situation if the soloist must be 50" from the microphone. A rug placed at the position of the floor bounce can be very effective in the upper audio frequency range. A super- or hyper-cardioid microphone could be used to reduce the reflected component another decibel or two. Thus our readily available remedies are quite limited, leaving the 50" distance to the microphone with basic problems unless a "shotgun" microphone is used which may well introduce a different set of problems.

Other close and distant microphone pickups are illustrated in Figure 9. Figure 9(B) shows a common geometry which can result in serious degradation of quality. Assume that d_1 = 12" and d_2 = 25". Then sound along the d_2 path would arrive close to 1 ms after d_1 which takes significant notches from the signal spectrum. the amplitude ratio would be close to 0.5 and, referring to Figure 5, we see that interference peaks would rise to about +3 dB and nulls would dip to about -6 dB giving overall response irregularities of about 9 dB. Closer talking will help reduce this as well as a good sponge rubber pad on the desk top.

Distant microphone pickups, such as in Figure 9(C) give distinct cancellation effects. By placing the microphone on the floor, d1 is made approximately equal to d_2 (Figure 9(D)) thus minimizing the comb filter effect. There are several stands and foam protectors designed to support the microphone very close to the floor surface. Another recent approach is the Pressure Zone Microphone of Ed Long and Ron Wickersham (Alembic, Inc.) enthusiastically promoted by Don Davis[3].

Figure 9c

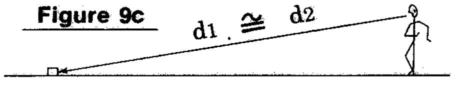
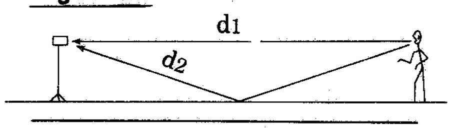


Figure 9d

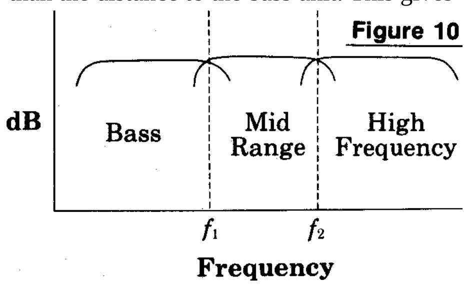


Example #4

Radiating the same signal from two separated loudspeakers or groups of loudspeakers lays down a comb filter pattern over the audience area[4]. On the line of symmetry between the two groups of radiators signals arrive at the same time and no comb filter effects are noticed, at least this could be true for one ear with the other ear plugged. Moving to either side of this line means that the auditor is closer to one group than another and delays generate the classical comb filter effect. In areas where one loudspeaker group is much stronger than the other due to path length differences and resulting inverse square fall off, the effects are modest. Directivity of the radiating sources also influences the area of the interference zone. Reverberation will also influence the detectability of the effect. With loudspeakers 25 feet apart, the 5 ms contours lie about 12" to either side of the line of symmetry down the center aisle as seen in Figure 4 - 5 of Reference 4.

Example #5

Multi-element loudspeakers can have their own private comb filter effects in the crossover region. In Figure 10 it is apparent that frequency f_1 is radiated by both the bass and midrange units, that they are of essentially equal amplitudes, and that the two radiators are not physically at the same point. This means that at a point in front of the loudspeaker the distance to the midrange unit may very well be different than the distance to the bass unit. This gives



all the ingredients for generation of comb filter perturbations of response in the crossover region. The same process is active at f_2 between the midrange and tweeter units. Actually a narrow band of frequencies is affected, the width limited by relative amplitudes of radiations from the two adjacent units. The steepness of the crossover curve determines the width of the frequency range affected. This is a highly complex problem that is being very actively studied at the present time and at least one monitor loudspeaker is being offered for sale claiming to minimize these defects (UREI 813)[3]. *(See editor's* note.)

How Audible Are Comb Filter Effects?

Just how serious a threat to the quality of our signal is this comb filter business, anyway? Psychoacoustical research on this subject must provide the answer to this question with any degree of finality. In the meantime subjective evaluations are all we have. Aside from its use for special effects, one inescapable conclusion is that comb filter effects certainly color the signal - but how much? For one thing, this depends on the position and depth of the nulls which, in turn, depend on the magnitude of the delay and the amplitude of the delayed signal component as compared to the direct. In general, comb filter effects give a "roughness" and unnatural "edge" to the signal. The critical bands of the human ear play a part in this as in all listening. The ear is a frequency analyzer with an analyzer bandwith (critical bands) of about 100 Hz below about 500 Hz. At 2 kHz the critical band is about 300 Hz wide and at 5 kHz about 900 Hz wide, etc. The 1/3 octave analyzer is a very, very rough approximation of the critical bands of the human ears. The point here is that at higher frequencies each critical band encompasses many peaks and nulls of commonly encountered comb filter responses. In Figure 2(C) the length of the lines labelled "CB" indicate roughly the width of the critical bands of the human ear at four frequencies. For example, at 10 kHz the critical bandwidth would include 3 peaks and associated nulls for the 1 ms delay case. As far as the response of the ear is concerned, these 3 peaks and nulls would not be delineated individually, but be averaged together in some way. On the other hand, for the 0.1 ms delay case of Figure 2(A) the peaks and nulls throughout the audible band would be well within the analyzing capability of the ear. It would seem, therefore, that the effects of the 0.1 ms comb filter on our signal quality would be much more apparent to the ear than the 1 ms comb filter.

Comb filter effects that are changing catch the attention more readily than unchanging ones. With two loudspeakers in the split system sound reinforcement, those seated in certain areas might be aware of considerable change in quality by moving the head. Walking down an aisle could add an undulating swishing to the program material radiated. In outdoor split loudspeaker setups refraction due to wind changes can introduce a variable swishing or a variable rough edge to the signal.

We can all be grateful that the application of the technique of time delay spectrometry introduced in 1967 by Heyser[6] brings acoustical comb filter effects out of the closet. OOO

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3 - Davis, Chips and Don Davis, *(LEDE) Live End - Dead End Control Room Acoustics, (TDS) Time Delay Spectrometry, (PZM) Pressure Zone Microphones*, Recording Engineer/Producer, Vol. 10, No. 1, February, 1979, p. 41.

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A Publication of the BAS

Speaker Cables Compared

Charles R. Ward, James E. Thompson,
and Mallory T. Harling

In the last few years there have been a number of new "high definition" speaker cables placed on the market. Reading the cable manufacturers' hoopla leads one to believe that using these new cables will make as big a difference in the sound of an audio system as buying new speakers. You will notice a lifting of the veil that has been concealing your music, more space around each instrument, and an increase in dynamic range-or so the claims go. Two articles by R. A. Greiner in the Boston Audio Society Speaker (December 1978, March 1979) presented electrical information on a variety of cables that cast some doubt on the claims made by the manufacturers of these special cables. The figures do not refute the subjective findings of many audiophiles that these cables sound different than more conventional cables, but they do indicate that some of these differences maybe due to cable-induced distortions.

The three important electrical parameters of eleven speaker cables are shown in Table 1. These data were compiled from figures reported by Greiner, manufacturers, and our own measurements. All of the cables use stranded wire construction, so skin effect should be negligible at frequencies in and beyond the audio range. Resistance values are shown as the sum of both leads of a single cable.

Table 1. Electrical Specifications of Speaker Cables

Cable	Inductance (μH/m)	Capacitance (pF/m)	Resistance (ohm/m)
No. 18	0.52	57	0.042
No. 16	0.59	50	0.026
No. 14	0.43	57	0.016
No. 12	0.39	74	0.010
Monster	0.40	66	0.0061
Lucas	0.63	48	0.010
Fulton (Gold)	0.32	86	0.0010
Polk	0.10	1630	0.026
Mogami	0.052	580	0.010
Discwasher	0.01	125	0.016
Audiosource	0.15	1990	0.021

Some manufacturers claim that speaker cables act as lossless transmission lines over the audio frequency range, and therefore, have tried to achieve an impedance value of approximately 8 ohms for their cables. The impedance (Z) for a lossless line operating at high frequencies is found by:

$$Z = (L/C)^{0.5} \quad (1)$$

However, this equation is appropriate only when $wL >> R$, where $w = 2\pi f$ and f is the frequency. When $wL << R$, then the impedance is found by:

$$Z = (R/wC)^{0.5} \quad (2)$$

For radio frequencies, the first equation is appropriate for determining Z, but in the audio range, the second equation yields more realistic results. However, it appears as though most cable manufacturers are using the first equation for their impedance calculations. We have, therefore, reported impedance values in Table 2 using equation (1). To achieve an impedance of 8 ohms using equation (1), the cable inductance must be lowered, or the capacitance raised, or both.

A lossless line is one in which all frequencies travel through the cable at the same velocity. The frequency (f_m) above which a cable begins to act as a lossless line can be calculated from:

$$f_m = R/2\pi L \quad (3)$$

Table 2. Transmission Line Parameters of Speaker Cables

Cable	Impedance (Z)	f_m (kHz)	Velocity (10^7 m/s) 50 Hz	Velocity (10^7 m/s) 15 kHz	10 m Cable ΔT (s)
No. 18	96	13	1.6	17	0.56
No. 16	109	7.0	2.2	18	0.40
No. 14	87	5.9	2.6	20	0.34
No. 12	90	4.1	2.9	18	0.29
Monster	80	2.4	3.9	19	0.20
Lucas	115	2.5	3.6	18	0.22
Fulton (Gold)	61	0.5	8.1	19	0.07
Polk	7.8	41	0.38	5.6	2.4
Mogami	9.5	31	1.0	14	0.93
Discwasher	8	300	1.8	30	0.52
Audiosource	8.6	23	0.39	4.9	2.4

Table 2 indicates that none of the cables acts as a lossless line over the entire audio band. At frequencies below f_m, the cable will act as a lossy line, one in which different frequencies travel at different velocities through the cable (frequency dispersion). The velocities of 50-Hz and 15-kHz signals as well as the time delay (ΔT) between the two frequencies in each cable are shown in Table 2. The velocities of various frequencies in the cables can be found by:

$$V = w/\beta \quad (4)$$

$$\beta = 0.71 \{wC[(R^2 + w^2L^2)^{0.5} + wL]\}^{0.5} \quad (5)$$

It is interesting to note that the most dispersive cables are the ones with the lowest impedance values.

The value of achieving an impedance match between the cable and speaker is reduced when one realizes that the cable, if it is to act as a transmission line with no reflections, must be terminated at both ends by its characteristic impedance. The impedance of most speakers changes constantly with frequency, and the output impedance of most amplifiers is well below 0.1 ohm. It should be obvious that speaker cables are never terminated by their characteristic impedance.

[In order for a cable to act as a transmission line, it must be many wavelengths long at the lowest frequency of interest. Audio signals traveling through speaker cable have wavelengths ranging between 9 and 9000 miles. Most people, including the authors, advise against using speaker cables this long. - Ed.]

Some of the special cables have very high capacitance. When capacitance is added to the output of most power amplifiers, it will cause them to ring; that is, to overshoot on square waves. The more capacitance, the more ringing there will be. As an example, the Hafler DH-200 amplifier exhibits a 15% overshoot on 10-kHz square waves when driving a 1-mFd capacitor in parallel with an 8-ohm load. This rises to 25% when the capacitance is 2 mFd. Many amps show much worse behavior with capacitive loads.

To summarize all the data, a good speaker cable should have:

a. Very low resistance. This is achieved by using heavy gauge cables-12 gauge or less. Since the internal wiring and contacts of most speakers approach a resistance of 0.1 ohm, the total speaker cable resistance should be kept to about half this value.

b. Low capacitance. The total capacitance of the cable should be kept below 1000 pF to prevent ringing and to minimize rise time in the cable.

c. Low inductance. The total inductance of the cable should be kept below 5 μH to prevent frequency dispersion in the cable. This is perhaps the least significant parameters, since the inductance of speakers may rise to as much as 10 pH at high frequencies.

It is possible to lower inductance without raising the capacitance by twisting the conductors at right angles to each other. An "ideal" cable might consist of 10-gauge stranded cable with the two conductors wrapped at 90°.

One factor should be apparent from this discussion. Keep the speaker cables short. All the factors affecting cable performance improve with shorter cables. The power amp should be located near the speakers and a high-quality phono cable used to connect the preamp and power amp. The characteristics for several phono interconnect cables are shown in Table 3. The total capacitance for the interconnect should be kept below 3000 pF to avoid high-frequency losses.

Table 3. Electrical Specifications of Coaxial Interconnects

Cable	Capacitance (pF/m)	Resistance (ohm/m)
Goldens	75	<0.2
Audio-Technica (610a)	66	<0.2
Belden 8421	55	<0.2
Columbia 1369	75	<0.3
Columbia 1389	55	<0.2
Sound Connectors	352	<0.1
Audio Authority	52	<0.2

Some of the performance differences noted for the "high-performance" speaker cables versus the more, usual speaker cables could be accounted for by the following:

a. The special cables are available in precut lengths which are usually shorter than the wires otherwise supplied in hi-fi stores. This may encourage purchasers to use shorter cables.

b. The special cables usually have solid, well-made connectors. This lowers the contact resistance in the amp -cable -speaker interface.

c. The special cables are usually a heavier gauge wire than most people use. The lower resistance of the wire, the more power is transferred, along with a reduction of the other problems mentioned earlier.

d. The special cables featuring high capacitance could be contaminating the signal by introducing high-frequency distortion. This might initially be mistaken for "better highs."

The first three items would lead to more gain in the amplification chain when using the special cables. Anyone familiar with comparative listening tests will know that "louder sounds better," especially when other sonic differences are non-existent.

In an attempt to collect data to test hypothesis (d), we set up the following experiment. A Hafler DH-200 power amp was connected to an 8-ohm dynamic speaker using an assortment of the cables shown in Table 1. A 10-kHz square wave was fed to the amp and monitored on an oscilloscope at the speaker input terminals. The following 14-ft lengths of cables were used: No. 16, No. 14, Monster, Polk (12 ft), Discwasher, Fulton Gold, and a twisted pair `of No. 8 wire.

The waveform obtained when the amp was driving a purely resistive load was compared to those obtained using the various 14-ft cables. <u>They were all exactly alike,</u> exhibiting a nearly perfect waveform with a 2.5-μs rise-time (10-90%). Even on an expanded range there were no differences to be observed. The same results were obtained with 20-Hz square waves. This is an interesting result when compared to claims made by Discwasher and Polk that regular parallel cables will not transmit a square wave.

We repeated the experiment using 30-ft lengths of the following cables: No. 14, Polk (24 ft), Discwasher, and Audio Source. This time, using the expanded range, there were clear differences among the various cables. The cables which use a special wrapping construction (Audio Source, Polk) showed distinct saw-tooth edges on the leading and trailing edges of the square wave. The high capacitance Audio Source cable showed the most distortion. The coaxial-design Discwasher cable was a little better than the Polk cable, but the <u>best of all was the No. 14 wire.</u> Representations of the waveforms are shown in Figure 1.

These data support our original contention that a speaker cable should be as large as possible to minimize resistive losses, constructed in such a way as to minimize capacitance, and as short as possible. Regular No. 12 or No. 10 stranded wire fits this description well.

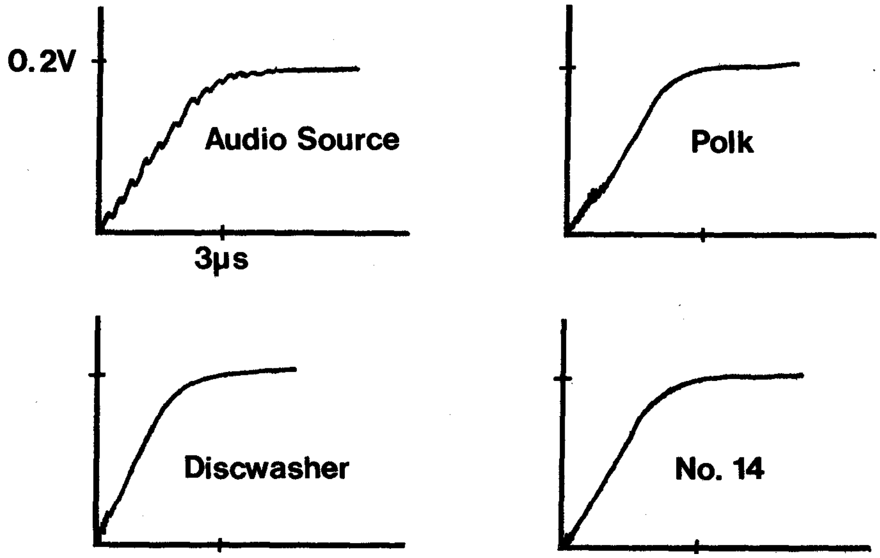

Figure 1 Leading Edge of 10kHz Square Wave