THE B.A.S. SPEAKER

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In This Issue

Those of us who enjoy listening with headphones usually have to put up with the relatively poor headphone output circuit in our amplifier or receiver. This is often only a high-value resistor in series with the main output that serves to cut the amplitude and protect the headphones. At the same time, this resistor presents a low damping factor to the headphones, which may adversely affect the frequency response. In this month's feature article, Peter Mitchell, with in-field monitoring in mind, describes two easy to build headphone amplifiers. As usual, Peter has included detailed assembly instructions and performance test data.

Most audiophiles absorb certain concepts with their mother's milk, it seems, then spend years unlearning them. Dan Shanefield who must be one of New Jersey's most experimentally minded audiophiles—appears in print again this month with an exploration of some of these concepts (he says he has about half a dozen more to discuss later). Some are new and controversial, like transient intermodulation (TIM) distortion; others are tried and perhaps untrue, like corner placement of loudspeakers for enhanced bass response.

We would welcome discussion of Dan's experiments, and note in passing that Dan not only has repeated the experiments on which he bases his deflation of these "audio myths," but has used different instruments and sometimes different measurement techniques to double and triple check his results. But if you disagree, say so, and we'll look into it.

There has been a lot of discussion about the audibility of phase distortion and it will probably continue so far as mid and high frequencies are concerned. But a moment's thought about the effect of low-frequency phase shift yields a frightening (and audible) conclusion. Imagine a 45-degree phase shift at 30 Hz. Since a complete cycle represents 360 degrees and takes 33⅓ milli-seconds to complete, 45 degrees represents a shift in time of just over 4 milliseconds. (Since sound travels at about 1.1 feet per millisecond, this is equivalent to a physical displacement of about 4 feet.) Now imagine a 30-Hz fundamental note with high harmonic content, and you can expect that you will hear the harmonics quite a bit earlier than the fundamental. In the real world every electronic circuit with a low-frequency rolloff produces phase shift in signals approaching this cutoff. Forty-five degrees is the actual phase shift of a single amplifier stage that is 3 dB down in response at that frequency. A drop of only 1 dB in response produces a shift of 27 degrees. It becomes obvious that flat low-frequency response takes on a new importance with this effect in mind. There are other significant contributions to bass delay. One is woofer inertia. Another is the shift in apparent tape recorder gap position at low frequencies.

Dennis Colin has isolated, manipulated, and subjectively evaluated bass phase or time skew distortion; his article on this phenomenon may rattle even those who think they have removed the last dB of imperfect reproduction from their music systems.

The Audio Amateur

The bonus coupon enclosed with last month's Speaker was dated for 1974 but is good also for a 1975 subscription. If anyone would like another coupon, send a stamped, self-addressed envelope to P.O. Box 7 and we will send you one.

Equipment for Sale

- Super-power amplifiers. Eight Dunlap-Clarke model 1000's (225 watts/channel at 8 ohms, 400-600 watts/channel at lower impedances). List $1200, for sale at $800. Two model 500 amplifiers (150 watts/channel at 8 ohms, 300 watts/channel at 4 ohms). List $800, for sale at $525. All units brand new, fully warranteed. Write Dunlap-Clarke Electronics, 230 Calvary St., Waltham, Mass. 02154, Attention: Ron Dunlap.
- Columbia Masterworks SQ decoder (simple type), $9.00. Call Ira Leonard, 729-5700 (days).
- Want to trade front panel Dyna PAS-3 for front panel Dyna PAS-2. Call Mark Saklad, 862-5500, ext. 7856 (days), 861-1659 (nights).

Speaker Impedance Measurements

If you were spurred to action by Ron Dunlap's ominous warnings of gross variations in many speaker impedances as a function of frequency, there are two simple methods for measuring the impedance of your own speakers. The result of both is a plot of absolute impedance versus frequency without phase information or reactive versus resistive components. So far as the instantaneous load on your amplifier is concerned, it is sufficient to know the absolute impedance. You will need an ac voltmeter or oscilloscope capable of resolving 20 millivolts, an audio oscillator, a few resistors, and some graph paper. The ideal graph paper for this and other audio plotting is four-cycle semilog. This paper has a linear scale along the short side and four cycles of logarithmic scale along the long side. The four cycles will allow a plot from 10 Hz through 100 kHz, and since most physical phenomena are logarithmic in nature, audio curves are most easily plotted and interpreted in this format.

The first method (Fig. 1) is the simplest, since the speaker impedance is effectively read directly from the meter or oscilloscope. The speaker power level is quite low during this test, as it is effectively driven from a low-current source. Since the speaker receives a constant current, its impedance may be read directly in terms of the voltage across it. Connect a 1000-ohm, 1% resistor between your speaker and the "hot" or nonground side of your power amplifier. Connect the oscillator to your amplifier input and set it for 1 kHz (1000 cycles). Adjust the level for an amplifier output, V_o, of 10 volts (rms if using a meter, peak-to-peak if using a scope—for simplicity in reading). Now read the voltage across your speaker, V_s. The current is very nearly 10 milliamps, so every 10 millivolts is equal to 1 ohm of speaker impedance. (You must stick to rms or peak-to-peak readings throughout.)

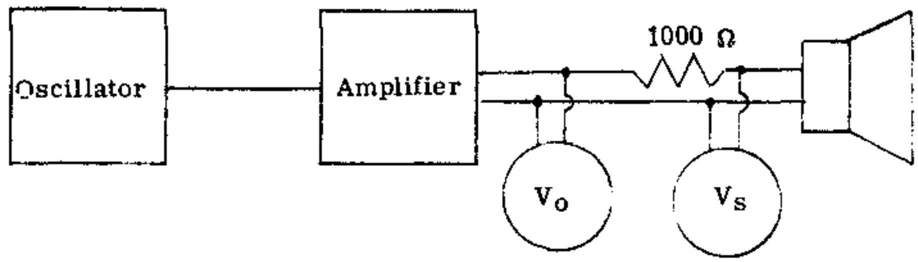


Fig. 1. Measuring speaker impedance—method 1

If you are unsure of your instruments' accuracy at these low levels, you can calibrate the whole setup and use a nonprecision 1000-ohm resistor by temporarily replacing the speaker with a known precision resistor in the range of 5 to 10 ohms and adjusting the amplifier output level until you get a reading on the meter or oscilloscope of 10 millivolts times the resistor value (e.g., 50 millivolts for 50 ohms). Note this level and maintain it throughout the test. Measure your speaker at 1 kHz and then make measurements at other points. It is accurate enough to assume a flat response for your amplifier so long as the tone controls are all set flat. If your oscillator also has a flat response, you may adjust the frequency and take readings without rechecking the amplifier output reading. If you are unsure, you should measure it as you test and keep it at the level set for the 1 kHz reading (10 volts or the value set up with the test resistor). You will get the most accurate plot with the least number of measurements if you adjust the frequency until you get a 1-ohm change, then note the frequency.

The second method (Fig. 2) tests speaker impedance under relatively high volume level conditions and does not read directly. In this method the speaker is driven at about 1 watt from a constant voltage and the current is measured as a small voltage across a low-value resistor. Connect a 1-ohm, 1% resistor between the speaker and amplifier ground. Connect the audio oscillator to the amplifier input and set the frequency at 1 kHz. Adjust the output level of the amplifier for a 100-millivolt reading across the resistor, V_r (rms if using a meter, peak-to-peak if using an oscilloscope). (Keep the meter or scope connections very close to the resistor.) This corresponds to a speaker current of 1/10 amp. Now measure the amplifier output voltage, V_o. The speaker

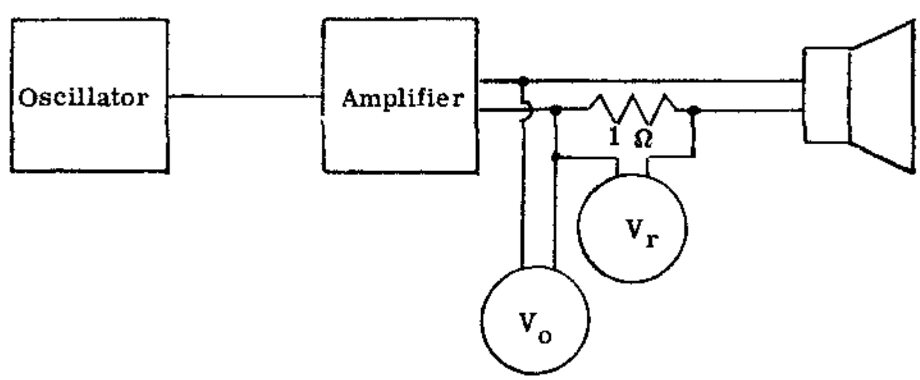


Fig. 2. Measuring speaker impedance—method 2

impedance in ohms is equal to the output level in volts minus 1.0 (for the resistor). For example, a reading of 9 volts (minus 1) indicates a speaker impedance of 8 ohms. Now, maintaining that output level, sweep the frequency, noting changes in the voltage across the 1-ohm resistor. It is most simple and accurate to adjust the frequency until you get a 10-millivolt change from the last recorded point, indicating a 10-percent change in total impedance. The resistance at each successive point is then

$$\frac{100\ \mathrm{mV}}{V_{res}} \times V_{out} - 1\ \mathrm{ohm}.$$

You may find an interesting similarity between your speaker impedance curve (Fig. 3) and the inverse of its frequency response curve. The 3-dB points on my Ohm F speakers match to within 1%. As the speaker impedance increases, it takes less power from the amplifier (which is a voltage source) and its output power will also decrease. I have heard arguments for driving speakers from a high impedance to partially counteract this effect. The ideal noninteractive

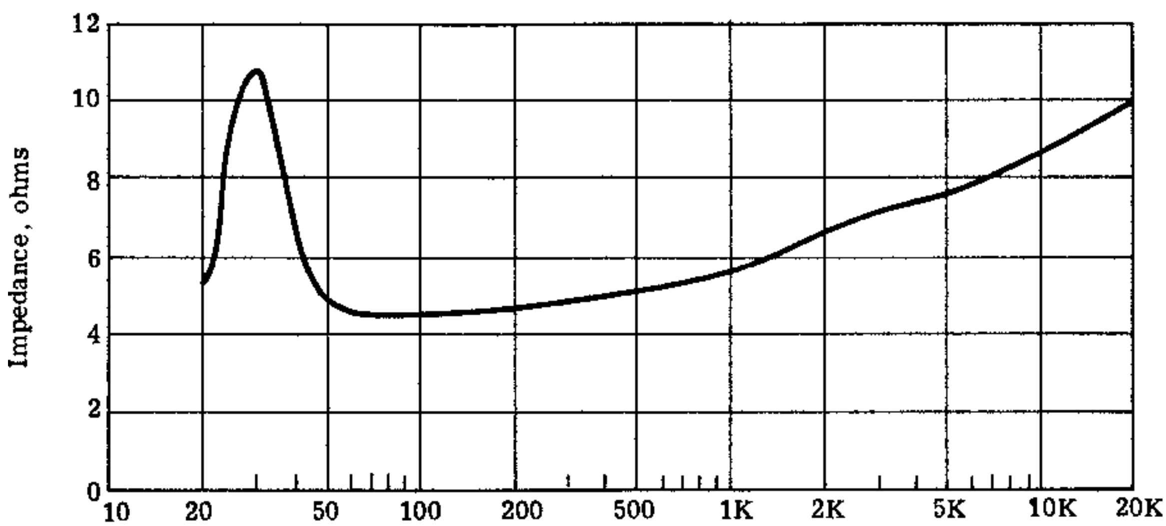


Fig. 3. Plot of Ohm F impedance using method 2

source would seem to be a damping factor of 1 or an amplifier output impedance equal to the nominal speaker impedance. If driven from a constant-current source, the power delivered to the speaker would actually increase as the speaker impedance increased. Unfortunately, an amplifier with an output impedance equal to the speaker impedance will put half its power into self-heating. This whole issue is complicated in multiple-driver speaker systems and by the efforts of manufacturers to overcome the effect. For the curious with lots of watts to spare, try putting a 50-watt, 4- to 8-ohm resistor in series with each speaker and judge or measure the effect in speaker frequency response of this matched impedance source.—Joel Cohen

Radio-Electronics Strikes Out

In the past Radio-Electronics magazine has not been strongly oriented toward hi-fi equipment, though hi-fi products and trends are included in its general coverage of electronic parts and products. R-E has now decided to get a piece of the action in the continually growing and mostly prosperous component hi-fi market. Starting in July they will be publishing full-bore test reports on audio components in each issue. The June issue contained a detailed summary of the tests to be performed on each kind of component. It was written by Len Feldman, who evidently will be in charge of the testing program; he is a consultant who has done some equipment reviews for Audio and is chairman of the IHF committee that is attempting to promulgate a new set of standards for FM tuner testing.

One is tempted to say, "Ho-hum, more test reports of limited usefulness; so what ?" But R-E invites closer attention. Feldman's article was designed to impress readers with the thoroughness of the planned test reports. The heading claims that "Our approach to testing is quite different from anything that has been done by other publications and is designed to be more informative." Furthermore, R-E is addressed to a readership of service technicians, engineers, and electronics hobbyists, so its equipment reviews ought to be able to discuss some of the generally ignored technical issues in current equipment designs, unencumbered by the need to translate everything for novices. Finally, the limited usefulness to audiophiles of the reviews in Stereo Review and High Fidelity is in part due to publisher pressure not to offend the advertisers, who provide the majority of each magazine's income. Larry Klein and Julian Hirsch, for instance, have a substantially more incisive understanding of the controversial issues in audio than you might guess from their published columns. Ed Dell's recent editorials in The Audio Amateur have cogently explored the problem of advertising influence on reviewing policies. In the case of Radio-Electronics, however, only a small fraction of the magazine's income comes from hi-fi manufacturers; most of the ads are for other electronic products, parts, and service aids.

So both the expectation of what R-E could do in hi-fi reviewing if it chooses to, and R-E's claim that its reports will be superior to those in other magazines, invite a close examination of Feldman's published test plans. Unfortunately that examination proves to be disappointing. While R-E's test reports will provide some useful information, there are gaping holes in the test plans, and it is evident that some of the essential distinguishing differences between competing components will be either ignored or deliberately concealed in R-E's reviews. Some specific examples follow.

FM Frequency Response. Rather than showing the actual response graph, R-E's reports will tabulate the "±X dB" deviation in the 50- to 15,000-Hz range. This approach hides essential information at both low and high frequencies.

Consider the low frequencies first. Contrary to popular myth, the FCC does not forbid the transmission of frequencies below 50 Hz; it simply requires that FM transmitters be flat down to 50 Hz, and response below that is optional. Some FM transmitters, as WGBH has demonstrated many times, are good to 30 Hz or below, and on a good broadcast a tuner with a low-end rolloff (such as an early production Dyna FM-5) sounds audibly inferior to a tuner having a truly flat low end. But R-E's reports will conceal such a difference.

At high frequencies there are two reasons why a tuner might not be perfectly flat to 15 kHz, and they have very different audible consequences which will be concealed by R-E's reports. One is the de-emphasis error, which will affect all frequencies above 2000 Hz and is plainly audible if the error amounts to 1 dB or more. The other is the low-pass filter required to attenuate the 19- and 38-kHz pilot tones; in many tuners this filter starts rolling off a little before 15 kHz, but that loss is generally inaudible if the response is otherwise flat (correct de-emphasis). A tuner that is 2 dB down at 15 kHz because of de-emphasis error will sound distinctly dull, while a tuner that is flat to 13 kHz and then down 2 dB at 15 kHz because of the pilot filter will sound fine; yet both tuners will be reported as measuring identically according to R-E's stated test report format.

FM Harmonic Distortion. R-E's tuner distortion tests were deliberately designed by Feldman to conceal one of the parameters that distinguish an excellent tuner from an ordinary one—namely the amount of intermodulation ("beat-note") distortion between high audio frequencies and the 19-kHz pilot in the stereo mode. This distortion shows up in High Fidelity's test reports as the "THD" figure at 10 kHz in stereo. (Genuine harmonic distortion cannot be measured or heard in tuners at frequencies above a few thousand hertz because, the harmonics are above the 15-kHz cutoff of the tuner. So any high-frequency `THD" number in stereo actually reflects IM between the audio tone and the pilot tone, generating non-harmonic audible garbage.) In many tuners if a 12-kHz tone is fed in, the multiplex decoder produces a 7-kHz IM distortion product that is subjectively louder than the 12-kHz fundamental. So far as I know, the first multiplex decoder that was reasonably free of this garbage was that in the classic Acoustic Research tuner and receiver. Only recently, with the advent of the Motorola 1310P phase-locked loop IC and similar devices, has a low level of high-frequency distortion in stereo become fairly widespread. However, the revised IHF tuner test standard, which Feldman's committee has developed, would conceal the differences among tuners in this regard, and R-E's tests follow that lead by limiting distortion measurements to a maximum frequency of 6000 Hz. There is no doubt that this is a conscious, deliberate choice. Feldman said so in the January 1974 Audio: "The lower figure is chosen for the highest frequency to be measured in stereo because many tuners with less than perfect multiplex decoding circuitry often produce sizable 'beats' between high-frequency modulating frequencies and internally generated 19-kHz and 38-kHz pilot and subcarrier signals." Of course if these beats were inaudible, it wouldn't matter, but our listening tests (both with a low-distortion FM signal generator and in tuner A-B comparisons on high-quality broadcasts) have shown the IM products from some multiplex decoders to be plainly audible.

Signal-to-Noise Ratio. R-E's measurements of S/N evidently will be unweighted, making them useful only for detecting grossly poor tuners and amplifiers. In most audio equipment, the measured noise consists as much of hum and subsonic transistor flicker noise as it does of hiss, and so the measured S/N values do not correlate with the product's audible noise. In general, all noise measurements should be bandpass-filtered to exclude inaudible contributions and then should be A weighted to conform to the ear's response. Once the weighted noise figure is supplied, the unweighted number may be useful as a supplement in order to indicate the potentially problematic presence of low-frequency or ultrasonic garbage in the signal. Incidentally, later this summer the BAS will publish the design for an active noise weighting filter.

Amplifier Power. R-E's amplifier power output measurements apparently will be made using only purely resistive loads, despite the generally agreed fact that audible differences among modern amplifiers usually relate to the response of the amplifier (or its internal protection circuitry) to the reactive impedance load of real loudspeakers. Furthermore, in an exercise of what appears to be sheer stupidity, R-E has chosen to adopt FTC preconditioning procedures in all power-output measurements—virtually guaranteeing that in most cases R-E will report badly erroneous power-output figures for 4-ohm loads. R-E's motivation in this is obscure. The FTC regulation applies only to power claims made by manufacturers and dealers, and it is the proper responsibility of any independent test lab to report a product's actual performance capabilities regardless of any misguided government regulation covering only the manufacturer's advertising. Perhaps Feldman's intent in using FTC preconditioning is to embarrass and goad hi-fi manufacturers into taking concerted action against the FTC, since except for Dynaco, most companies have been scandalously timorous in their response to the FTC's mistake. In any case, for the benefit of consumers who may rely on R-E's reports, I hope that each amplifier's actual power output capabilities will be mentioned in a footnote somewhere in each report.

Tape Decks. R-E claims that each open-reel deck will be tested with both standard and Cr02 tape. This is silly, of course, since CrO_2 provides no significant benefit above 3¾ ips, and open-reel decks are not provided with the bias and equalization that CrO_2 would require.

R-E evidently plans to omit any mention of the input-overload levels in tape recorders—an important parameter in both mike and line inputs. S/N measurements evidently will be unweighted, making them about as useless as those in High Fidelity. Further complicating the issue, S/N values will be referenced only to the 0 VU point on the recorder's meters. The problem is that the recorded flux level corresponding to 0 VU varies drastically among manufacturers because of their choice of meter calibration. Consequently, for instance, if Technics and Tandberg cassette decks actually had the same real noise level (e.g., -60 dB relative to the standard Dolby cal level), R-E would describe the Tandberg as being 6 dB quieter—simply because Technics uses meters on which 0 VU is 3 dB below the standard flux level of 200 nanowebers/meter, while Tandberg uses peak-reading meters whose "0" is 3 dB above 200 nW/m.—Peter Mitchell

A-B Testing Revisited

When A-B'ing components, I have found it useful to first "rough" match the relative levels as closely as possible by ear and then make the "fine" adjustment while listening for—and attempting to eliminate—quality differences. I have proven to myself repeatedly that level differences too small to be heard as such are frequently heard instead as quality differences. When the relative levels are adjusted so that the quality differences disappear (or are very small), the ear does not hear a level difference.

This is not to say there are never objective and audible differences among equipment. But I'm convinced that the vast majority of those listeners, reviewers, critics, etc., who consistently hear distinct quality differences between power amplifiers are really responding to those minute level differences. The test for this is easy; simply slowly raise the level of the "inferior" amplifier by small increments while A-B'ing. If at some level its sound becomes indistinguishable from that of the "superior" amplifier, and yet does not appear to be any louder, my point is proven. Three caveats: both amplifiers should, of course, be tested in advance for normal performance, they should be driving conventional speaker loads, and most important, they should be monitored for clipping. (Clipping, if not too severe, will be heard as a loss of dynamic range before it is audible as IM or THD.) Incidentally, I suspect this phenomenon results from something other than the ear's standard Fletcher-Munson response, but I have no idea what.

On this last point, Daniel Shanefield's reference (March Speaker, p. b4) to Julian Hirsch as insisting that the ESS Model 200 is audibly indistinguishable from more powerful amplifiers is not Julian's complete statement. . . . in the same sentence he goes on to say "so long as we did not exceed its maximum power capabilities." I think you'll agree that it's important to put Hirsch's statement in context; otherwise he appears to be holding a totally irrational view—and God knows there is too much of that already in the audio field.

I've been wanting to tell the people who put together The BAS Speaker how impressed I am with the publication. Unlike those who publish many of the other non-commercial publications, your group seems eminently sane in their approach to the audio art/science. —Larry Klein (Technical Editor, Stereo Review)

An Embarrassingly Simple Method of Tonearm Damping

My Rabco SL-8E tonearm, which, despite major weight-reduction surgery, still caused an ADC-26 cartridge to show visible cantilever motion almost constantly, seemed an obvious candidate for damping. I regarded the methods described in the Speaker as poorly suited for the straight-line arm, unesthetic, and too much work.

The search for a solution, based on a desire to be as unoriginal as possible (i.e., how do the manufacturers do it?), led to Audio Distributors in Grand Rapids, Michigan, where I purchased for $1.50 a two-gram tube of the viscous silicone fluid used in the Gray Micro-Trak damped tonearm. (It ought to be available in other places.) I applied, with a toothpick, a small amount of the gel-like substance directly to the horizontal bearings of the tonearm. The fluid is sufficiently viscous and adhesive that it doesn't go anywhere, but infiltrates a bearing and sits in a small glob around it, indefinitely.

Tracking has unquestionably improved. The cartridge cantilever now shows motion only on the most violent record irregularities. Having failed to make careful before-and-after listening tests, I cannot be specific, but feel that there has been some improvement in the sound.—Jack Stevens

Jack says that this may be the first of a series of notes along the lines of "Dr. Straightline"; or, "How I Learned to Stop Worrying and Love the Arm." I'm happy: I think it's about time something in this hobby became embarrassingly simple.—Ed.

Capacitance and Your Phono Cartridge

The upper-midrange and high-frequency response of a magnetic phono cartridge is affected by the capacitance to which the cartridge is connected. The best known case of this was the Shure V-15 type II, whose sensitivity to capacitance was demonstrated at the October 1972 BAS meeting. Most manufacturers have been rather reticent about this problem, and the glossy hi-fi magazines have failed to fill this information gap. So Dynaco took the initiative and canvassed the manufacturers of cartridges and turntables sold in the USA, and published the resulting data in the owner's manual for the PAT-5 preamp. We have gratefully lifted the following tables from the PAT-5 manual. The first table lists the recommended load capacitance for the cartridge. The second table gives the total capacitance of the turntable's arm wiring and signal cables. Some manufacturers, listed in the third table, provided only the capacitance of the arm wiring, omitting the signal cables entirely, so their data is given in parentheses to indicate that you must add the cable capacitance. Ordinary shielded audio cable typically has a capacitance of about 30 pico-farads (pF) per foot, yielding about 90 pF for a 3-foot cable. Finally, to compute the total capaci-tance, you must include the phono preamp, which in most cases will necessitate an inquiry to the manufacturer of your amp. Fisher amplifiers commonly have about 300 pF at their phono inputs to suppress RFI; the AR amp's phono input is about 50 pF, and Dyna specifies less than 10 pF for the PAT-5.

If any members have conflicting or additional data to add to the following, we would be pleased to publish it in a future issue.

Cartridge	Recommended Capacitance
Audio-Technica (all models)	100 pF
ADC (all stereo models)	Less than 400 pF (no optimum)
B&O MMC-6000	100 pF
Decca (all stereo models)	No optimum
Empire	
(all stereo models)	150 pF
(all quad models)	100 pF

Cartridge	Recommended Capacitance
Ortofon	
(M15/F15 series)	400 pF
(SL15 series)	Unaffected
Pickering	
(all stereo models)	255 pF
(all quad models)	100 pF
Shure (all stereo models)	450 pF
Stanton	
(all stereo models)	255 pF
(all quad models)	100 pF

Turntable	Capacitance of Arm + Cable
AR (all models)	135 pF
BSR 810/710	285 pF
620/610	210 pF
Empire (all models)	110 pF
Glenburn 2155 A/Q	90 pF
2155A/215 5S	330 pF
Hitachi PS-12	80 pF

Turntable	Capacitance (Arm Only)
BIC 960/980	(125 pF)
Decca arm	(300 pF)
Dual (all new models)	(70 pF max.)
PE (all new models)	(20 pF)
Rabco (all models)	(68 pF)
Sanyo TP 805A/805B	(55 pF)
Shure SME arm	(130 pF)
Technics SL1300/1200	(56 pF)
SL1100	(79 pF)
Thorens (all models)	(90 pF)

TE 814 Notes

814 Equalizer Update

In the April issue I suggested that since the desired boost curve in the 814 equalizer is rather broad, the exact peak frequency is not critical, and any capacitors in the range from 0.004 to 0.006 microfarad would be suitable. I have built two 814 equalizers, one using 0.0047-μ F capacitors (yielding a peak frequency of about 3900 Hz) and one using 0.0056-μF values (providing maximum boost at about 3200 Hz). In A-B tests, the equalizer containing 0.0056- μF capacitors sounds better.—Peter Mitchell

Windscreens for the 814 Microphone

It is desirable to keep a foam windscreen permanently installed on the 814 or any good microphone. In addition to minimizing wind noise and breath pops, it also shields the microphone element from dust, protects it from handling damage, and cushions it when dropped. Alan Southwick suggested an excellent windscreen, the one which AKG supplies for their C-451 mike, but it costs $6. Looking for a less expensive alternative, Ira Leonard noted the Olson MK-085 at under $2. In a test both the Olson and AKG windscreens proved effective. But whereas the AKG unit absorbs less than 1 dB of the sound at 10,000 Hz, the Olson windscreen attenuates the incoming sound by about 4 dB at that frequency and produces a subjectively dull sound with the 814 mike. So the search for a high-quality, low-cost windscreen continues.—Peter Mitchell

The Dangerous Loudspeaker

Can a loudspeaker safely be placed on top of a cabinet in which magnetic tapes are stored? To find out, I measured the external magnetic field of a few loudspeakers, using a magnetometer. The result was a slightly frightening confirmation of the old warning that your loudspeakers will cheerfully ruin your tapes if you are not careful. For example, a tape placed at a random location on the exterior cabinet surface of a Smaller Advent, AR-3a, or AR-LST will be exposed to a magnetic field of 10 to 20 gauss, and there are some locations on the 3a and LST where the field strength approaches 50 gauss! For comparison, the maximum safe ambient magnetic field for a recorded tape is about 1 gauss, and the earth has a permanent field of about 0.5 gauss.

But you cannot use a speaker's magnetic field to bulk-erase tapes. It selectively erases just the highs, leaving low and middle frequencies largely unaltered. And since it is a dc field, it also tends to add noise and distortion to the recording.

Fortunately the inverse-square law operates with magnetic fields, so the field strength decreases rapidly as you move away from the loudspeaker. At a distance of 10 to 20 inches from each speaker, depending on direction, the field drops to about 1 gauss. So as a general rule, in order to be safe, always keep magnetic tapes at least 2 feet away from the exterior surface of any loudspeaker.—Peter Mitchell

More on the Cartrivision Video Tape Recorder

I have had my Cartrivision VTR for over a month and have spent close to 50 hours in some necessary and much needless fiddling with it. The design and quality of manufacture are impressive. The performance is splendid, even when viewed on my 7-foot Advent TV screen. They are still available from at least three sources. I got mine from Knickerbocker Enterprises, 114 Windcliffe Drive, Ballwin, Mo. 63011. Their price is $195, but they earlier agreed to sell to BAS members at $150. Media Associates, 1470 North Fourth Street, San Jose, Ca. 95112, is staffed by ex-employees of Cartrivision and sells a unit that is completely recalibrated, in a cabinet with all the knobs, etc., plus an RF output set for channel 3 at $365. Olson Division of Teledyne has factory checked-out units at $300, as is. Olson also has the one-half front panel plus knob set from Packard-Bell at $25. Tape is available on a catch-as-catch-can basis from all three sources. Knickerbocker has the best price as long as his supply lasts: 1/2-hour cartridges are $6; 1 hour, $12; 1 hour 54 minutes, $22. Media Associates charges $19.40, $24.60, and $32.56, respectively, and Olson charges $22.98 for 1/2 hour only. All three sources also have the high quality zoom lens B/W camera for $175 to $200. It is self-contained except for dc power.

The VTR is a helical scan skip field unit. There are three playback heads, but only one is used to record every third field. Since there are 60 fields per second, this unit records at a rate of 20 fields per second, alternating between the first and second portion of each complete frame. The only noticeable result is a slight jerky motion with some action in films, since they are already being unevenly synchronized with the TV frame rate (6 times a second a film frame is scanned twice). The video signal-to-noise ratio is specified at over 40 dB, and indeed it looks very good. It is achieved by converting the luminance signal into "digital" pulses. The luminance and chroma are separated. The luminance drives a 3.5- to 5-MHz voltage-controlled oscillator that puts out square waves, so the signal is either there or not and is unaffected by variations in amplitude due to tape variances or noise. The chroma is added back after heterodyning it to a lower frequency (0.1 to 1.1 MHz) and rides as an analog signal on top of the pulses. On playback, the original signal is reconstructed and a dropout compensator takes care of most streaks due to dust or tape scratches. This is a circuit that delays the composite video output signal by 63 microseconds (the time of one horizontal line) then detects loss of the main (undelayed) output signal and switches to the delayed signal until the level returns. Thus it inserts a piece of the line above the one being played to fill in the hole which otherwise would look like a horizontal black streak. Several members can attest to the resultant quality.

The sound is also good. In fact it is set up for stereo playback, since some of the planned prerecorded tapes were to have dual language or stereo sound tracks. In record mode, the two heads are driven together but could easily be separated to record things like BSO simulcasts. Tape speed is 3.8 ips, and without trimming the response goes up to 10 kHz.

There is a built-in timer to turn the VTR and TV on after up to 8 hours delay and off again when the tape runs out.

The VTR has no RF input or output provisions, so you must tap into the video amplifier and audio amplifier in your TV. This is not tricky for the knowledgeable, but for those in awe of cutting into their sets, it may be a drawback. Media Associates sells an RF output unit for $79, and you may be able to make a tuner out of a good B/W TV set (the color signal is there even if the set doesn't use it so long as the IF bandwidth is high enough). For that matter, this unit is so cheap, you could buy a second color set just for it and still spend less than half the cost of the next lowest priced color VTR.

A recent rundown of a new Sony cassette VTR indicates they have copied a number of Cartrivision's techniques, including the luminance pulse conversion.—Joel Cohen

Recording Highs and Lows

Here from Dow Williams all the way out in Salinas, California, are four recommendations on budget labels no less.

- Crumb/"Makrokosmos," Vol. 1/Nonesuch H-71293/Sonic delights galore. A sharp, clean recording with jolting peaks. The only record I have which overloads the amps (or it may be pickup mistracking). At one point during the first half of side 1 there is a horrendous piano chord that produces a clacking noise in the speakers. Sheffield Vol. III on the other hand, remains clean throughout.
- "A Festival of Trumpets"/Nonesuch H-71301/If music can be delicious, this is ambrosia. Well balanced harpsichord. Compare with Vivaldi, Nonesuch H-71022, for thin harpsichord (otherwise OK).
- Hayden/"Duo Concertante for Viola and Organ"/Argo ZRG 631/You will play this one to death in a month's time; better get two copies.
- Stravinsky/"Pulcinella"/Argo ZRG 575/When I sit down to write something about this record, I have to get up and play it again—such is the magnetic appeal of the music. The recording is first class.

In the Literature

In response to numerous requests for subscription information on the magazines listed in this column, we include the following addresses and subscription rates. The rates listed are those published in the magazines; lower rates are sometimes available through coupons or promotions.

- Acoustical Society of America, Journal of, Subscription Fulfillment Division, American Institute of Physics, 335 East 45th Street, New York, N.Y. 10017. Rates: $30/year (better check the local university library).
- Audio, 134 North 13th Street, Philadelphia, Pa. 19107. Rates: $7/year, $12/2 years, $17/3 years (but look for $3.99/year specials).
- Audio Amateur, P.O. Box 30, Swarthmore, Pa. 19081. Rates: $7/year.
- Audio Scene/Canada, 481 University Avenue, Toronto M5W1A7, Canada. Rates: U.S. $10/year, Canada $8/year.
- IEEE Transactions on Acoustics, Speech and Signal Processing, IEEE, 345 East 47th Street, New York, N.Y. 10017. Rates to nonmenbers available on request.
- Popular Electronics, P.O. Box 2774, Denver, Colo. 80302. Rates: $6.98/year.
- Radio Electronics Service, Boulder, Colo. 80302. Rates: $8.75/year.
- The Stereophile, P.O. Box 49, Elwyn, Pa. 10963. Rates: $7/year.
- Wireless World, Oakfield House, Perry Mount Road, Haywards Heath, Sussex RH16 3DA, England. Rates: $15.60/year, $ 34.80/3 years, half price to students.

Acoustical Society of America, Journal of, May 1975

- Two Channel Listening to Musical Tones: This one is really far out. "Subjects listened to a dichotic tonal sequence consisting of repetitive presentation of the C major scale with successive tones alternating from ear to ear. The scale was presented simultaneously in both ascending and descending form, such that when a component of the ascending scale was in one ear, a component of the descending scale was in the other. . . . Right-handers tended to perceive the upper tones . . . as emanating from the right earphone and the lower tones from the left .. "even after the earphones were switched. The results are pretty spooky, even worse than the strange sound "in the middle of our heads" found when listening through phones. Should we be testing speakers in rooms tuned for right-handed and left-handed listeners? (p. 1156)
- Relations Among Temporal Resolution, Forward Masking, and Simultaneous Masking: Also far out, and written in scientese, but measures our ability to discern two time-separated noise bursts of varying intensity, and also masking of a tone by a preceding burst of wide-band noise. Takes a lot of reading, but this type of article can help us understand exactly how we hear (p. 1169).

Audio, July 1975

- A skinny issue but dedicated to hard to find information about car radios. In addition to the usual "Audio"-type tabulated data, five units are tested in depth, and one (the incredible ADS 2001 amplifier and speaker system) gets a full Heyser review.

Audio Times, USA, April 15, 1975

- This is a dealer journal not available to individuals. Some items of interest are: Superscope Sales for 1974 were $157M, up $39M from 1973 (and this was a bad year ?): as we all know by now, sales of top-of-the-line items are still strong, with no middle or bottom; FTC now requires tags on loudspeakers giving the composition of the "wood"; a horrible letter from the Pickering sales manager compares the QDC-1 to a $1.79 crystal cartridge; a discussion of Audio Technica's method of mounting a square-shank stylus in a square hole (rather than a round one in a round hole) to keep alignment accurate; Advent is looking for sales people starting at $12.5K.

Audio Times, USA, May 1, 1975

- Items of interest include word that Hartley intends to push speakers even harder in the moderately priced markets; BASF sales are up to $517M from $379M last year; KLH is reorganizing its dealers, and the Research X line will be selectively handled and fair traded; and Fuji, of film and LED-readout-camera fame, is entering the magnetic tape market.

Consumer Electronics, May 1975

- Another dealer rag, this one with a much broader scope. In this issue, Connecticut, New York, and New Jersey have almost abolished their fair-trade laws; Sony has a crystal-controlled $580 turntable and a $800 speaker system; Ferrichrome switching will appear on most new cassette units, and front-loading will be more common; TEAC now has 25 models of their open-reel decks, up to $1500; and prices will jump just as soon as the economy can bear it, so buy now if you have the need and the cash.

EDN, June 5, 1975

- Today's Power Transistors Provide Prodigious Performance: A general review of power units, with a short discussion of why triple-epitaxial units are superior to triple-diffused units.

IEEE Transactions on Consumer Electronics, May 1975

- An Infrared Wireless Speaker System Utilizing a Super Wideband FM Carrier: Zenith has a design, very complex and advanced, for transmitting signals to the rear speaker/amplifiers (p. 115).

Popular Electronics, July 1975

- Choosing Your FM Antenna (p. 16)

Radio-Electronics, July 1975

- R-E's first in-depth test report, Sansui QRX6001 receiver (see Peter Mitchell's comments elsewhere in this issue.)
- Understanding Op Amps, Part II: A continuation of Don Lancaster's excellent series.
- All About Oscilloscopes, Part I: Beginning of a series by a Heathkit design engineer.

Wireless World, May 1975

- Audio Engineering Society's 50th Convention; report that the new Quad 100-watt-per-channel amp will be fully class A at low power, switching to push-pull at high levels; several notes on loudspeaker papers presented at the conference, with a figure of speaker output versus frequency and time which does Heyser one better; a promise that EMI will be upgrading the duplication of their cassettes; and the address to write to for a copy of the conference proceedings (p. 207).
- 75 Years of Magnetic Recording, Part 3.
- Noise, Part 3.
- Wireless World Dolby Reducer, Part 1: This is a superb introduction to the process of noise reduction, broader in scope and deeper in detail than any I've seen. The construction article of the unit will follow next month; this month only photographs of the (beautiful) IC unit were given, but without the schematic.

May BAS Meeting

Business and Open Discussion

Approximately 60 members met at GTE Labs on May 18. Jim Brinton described the BAS FM tuner clinics and accepted reservations for clinic appointments. Ira Leonard supplied a group purchase of BASF LP-35LH premium tape at $3.75 per reel. Al Foster accepted orders for a bulk purchase of Maxell UD-35, BASF CrO_2 cassettes, Sheffield Vol. III, and Shure and CBS test records. Ira Leonard also has for sale at only 75 cents each a large quantity of low-voltage power supplies (Radio Shack 12-704, 110 Vac in, 4.5 Vdc out at up to 60 mA). Fred Parmenter provided copies of the mail-order catalog of the Soundd Investment Co. (Box 338, Dunwoody, Ga. 30338); SI sells bulk quantities of tape at very attractive prices and has been the source of the BAS purchases of 3M 177 via Jim Richardson.

Peter Mitchell described a statistical pattern called the "BAS Officer Syndrome": each BAS officer in the history of the BAS has, at the time of his or her election, been the possessor of a medium-grade stereo system, and in turn each has ultimately acquired at bargain prices a set of

AR LST's and a Phase Linear 700). This curious statistical pattern, it was suggested, might serve as an incentive for a larger proportion of members to become actively involved in BAS affairs. He then announced the discovery of a new syndrome, namely that the members who write the most for the Speaker ultimately find themselves invited to write for the large national magazines; the current manifestation of this is Jim Brinton's cover article on tone arm damping in the July High Fidelity. So members should feel encouraged to write for the BAS Speaker—it may lead to larger things.

Meeting Feature: Ron Dunlap

The principal speaker at the meeting was the subject of a feature profile in this spring's Boston Phoenix hi-fi supplement. Ron Dunlap, who together with engineer Mel Clarke founded Dunlap-Clarke Electronics, spoke on the relationship between loudspeaker impedance characteristics and power-amplifier design requirements. Ron is a vigorous young entrepreneur who first became interested in amplifier design in 1968 while a medical student. Since then he has pursued two simultaneous careers as a physician and an amplifier manufacturer. He learned amp design largely by trial and error, beginning by obtaining the manual for a Scott amp, building a copy of that amp from spare parts, and then modifying it experimentally to further improve its performance. He later repeated the same trick with the schematic for a Crown DC-300. The success of those ventures served as the foundation for a continuing examination of design practices in modern amplifiers of various brands, while he was working part-time at an electronics lab where the availability of parts and test gear made experimenting easy.

To place amplifier output-stage design into perspective, Dunlap began his BAS talk by describing some of the common design approaches. In each of the accompanying schematics the transistors in the upper half of the diagram amplify the positive half of the waveform and the lower transistors carry the negative half, with the resulting currents combining on the center rail of the schematic (which goes to the speaker terminal). Figure 1 shows a design which was developed by RCA engineers in the mid-1960's but was not widely used (by Harman-Kardon in the Citation 12, for instance) until the cost of the required transistors descended to a reasonable level. Figure 2, from a design first published in 1969, is much better from a designer's point of view because it is a true "complementary" circuit. This means that the positive and negative halves of the circuit are genuine mirror images of each other, with the positive currents carried by npn devices (which have the emitter arrow pointing out of the transistor symbol) and the negative currents carried entirely by pnp's (with the arrow pointing in). Because of its symmetry, a complementary circuit tends to be freer of distortion and instability, but the required pnp devices have generally been available only in a limited variety and at much higher cost than npn's. So most designers have adopted "quasi-complementary" designs based on Fig. 1, in which npn devices are used for both the positive and negative sides of the circuit and additional components (symbolized by an "X" in the diagram) are used to trick the negative side into behaving as if it were made of pnp's.

Design attitudes were drastically altered by "whiz-kid" Bob Carver's introduction of the Phase Linear 700, in which he called attention to the importance of high instantaneous transients in the reproduction of music. Recall that in the Dyna Stereo 120, for instance, a strictly regulated 70-volt power supply limits not only rms but also peak-power outputs to about 60 watts, so that in music with a realistic peak-to-average ratio, the maximum rms level must be kept to 10 or 15 watts in order not to clip off the instantaneous peaks. The Phase 700 employed a "soft" high-voltage supply permitting a momentary output swing of nearly 200 volts, enabling the production of peak power outputs of several hundred watts. However, the lack of suitable high-power pnp's made a truly complementary circuit impractical at that level, so the 700's design (Fig. 3) is quasi-complementary, essentially a high-voltage adaptation of the Crown DC-300. The driver transistors are stacked up in series in order to handle the high signal voltages, and the parallel lines in the diagram indicate that multiple output transistors (all npn) are added in parallel to carry the currents involved.

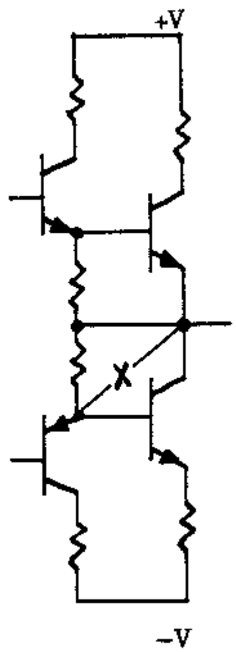


Fig. 1. Quasi-complementary circuit used in Harman-Kardon Citation 12

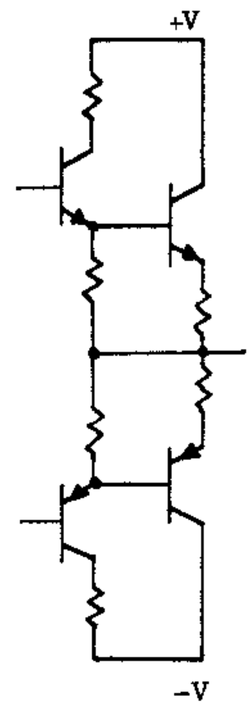


Fig. 2. True complementary circuit used by JBL and Accuphase

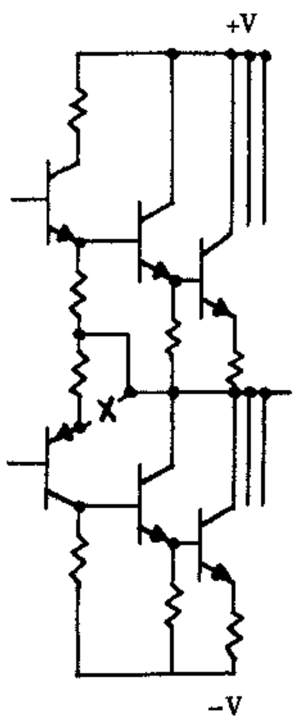


Fig. 3. Quasi-complementary circuit used by Phase Linear, Crown, and BGW

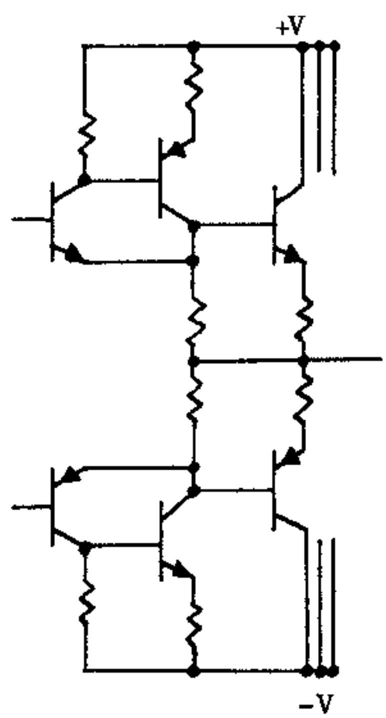


Fig. 4. True complementary circuit used by Dunlap-Clarke and Marantz types 15 and 16 (Ampzilla and Dyna Stereo 400 are similar but multiple output transistors are added in series)

Dunlap-Clarke adopted a slightly different perspective in designing their Dreadnaught amplifiers. Ron Dunlap contends that an amplifier's ability to deliver high peak currents to a load is as important as—perhaps more important than—the ability to produce high voltages. Ron mentioned two reasons for this: (1) the increasingly widespread use of double or multiple loudspeakers in parallel, and (2) the fact that many loudspeakers are highly reactive rather than resistive, meaning that they draw higher peak currents than one would expect from their nominal impedances:

Dunlap's circuit is shown in Fig. 4. The reduction of the maximum voltage to 160 volts (±80 volts), together with progress in the manufacture of pnp transistors, enabled a true complementary-symmetry approach, analogous in layout to the old Marantz 16 but capable of higher output levels and faster slew rates. The Dreadnaught 1000 has a rated output of 250 watts per channel into 8 ohms and 500 watts into 4 ohms, and in tests Ron says it has delivered 850 watts per channel into a 2-ohm load. The key to this is that the amplifier can deliver peak currents of 28 amperes to the load, nearly twice that of a Phase Linear 700.

The recent FTC directive on power ratings, Dunlap noted, will make it more difficult than ever for the consumer to identify those amplifiers which have a desirably high current-output capability. The ability of an amp to deliver high current to a load is synonymous with its ability to operate successfully with very low impedances, and one way to identify amps which are relatively free of current-limiting is to compare the 4-ohm and 8-ohm power ratings. If an amp can deliver 50% to 100% more power at 4 ohms than at 8, it probably can deliver current in a relatively uninhibited fashion. But an amp which can deliver only the same power at 4 ohms as at 8 ohms obviously is encountering current-limiting. The problem for the consumer is that this comparison may soon become impossible as 4-ohm ratings are tending to disappear entirely; most amps, regardless of their ability to pass the FTC preconditioning test at 8 ohms, cannot pass it at 4. Another method, admittedly crude, whereby you can estimate an amplifier's current capability is to examine it with its cover removed; the larger the power transformer and main filter capacitors, the greater the unit's output capability.

To support their view of the importance of current-output capability, Dunlap and Clarke have studied the current demands of loudspeakers through the use of a Hewlett-Packard vector impedance meter, a device which generates a small signal of any desired frequency, monitors both the voltage and current flow, and displays the phase and magnitude of the impedance (the ratio of voltage to current). In the near future Ron may publish the apparent impedance data obtained for various loudspeakers. For instance, multiple KLH-9's, which are notoriously amplifier-sensitive, were found to absorb very large amounts of current at only moderate drive voltages. In testing an AR LST, the instrument indicated an effective impedance of 6 to 8 ohms over most of the audio frequency range except in the bass; below 100 Hz the apparent impedance was said to be about 2 ohms, descending toward a short circuit below the audible range. (The autotransformer in the LST is effectively a short-circuit at dc, rising to a respectable impedance in the audio range; but because this reportedly caused failure in some amplifiers, current LST's contain a large input capacitor.)

Low reactive impedance is not the sole problem observed with the vector impedance meter: some speakers were found to have very high apparent impedance, as much as 30 to 50 ohms at some frequencies. The difficulty here is that if an amp attempts to deliver full power at high frequencies into a high load impedance, as it may when it is driven into clipping, an output transistor may go into "common-mode conduction" (in which it conducts both halves of the waveform instead of just one polarity as it should). Destruction of the transistor is a likely result, especially if a blown speaker-line fuse causes the amp to try to deliver full power to an infinite load impedance.

Unlike many amplifiers which have been designed entirely in the laboratory with 8-ohm pure-resistance loads, the Dreadnaught amps were designed largely on an empirical basis, i.e., by trial and error using real loudspeakers. The evolution of the elaborate protection circuitry illustrates this point best, particularly as it is generally agreed by manufacturers of super-power amps that audible misbehavior of an amp usually is due to the protection circuits rather than to the signal-handling circuits. Ron described the first step in this development as the adoption of basic short-circuit immunity, to prevent the output transistors from burning out when the user accidentally short-circuits a speaker line "hot" to ground. Next, since the amp can deliver high peak currents, the transistors must be protected from trying to deliver too much current under abnormal conditions; so a logic circuit continually monitors output current and voltage and shuts the amp down if the amp is continuously overdriven or if the load impedance is below 2 ohms. (The fact that the amp shuts down is a significant choice. Many amp protection circuits simply feed corrective signals to the input stages, altering the waveform to safe proportions and distorting the sound in the process. Dunlap and Clarke prefer protection circuits which never alter the sound and which, if activated, shut down the amp to notify the user that he or she is abusing it.) Next, the frequency response characteristic of the protection sensor was modified to prevent the amplification of large dc transients such as a dropped tone arm or a severe switching pop. Further experiments indicated that the protection circuit would inhibit the amp from delivering full power into some of the highly reactive speakers on the market (because of the phase shift between voltage and current), making more modification necessary. Ron claimed that the amp now can deliver full power into any reactive speaker without blowing up either the speaker or its own transistors. As a final step, speaker-line fuses are provided so that the user can choose the maximum continuous power level that is safe for his speakers—without, of course, limiting the instantaneous peak-power reproduction that is essential for lifelike sound. (In this regard note the demonstration with the Smaller Advent speaker described below.)

One of the beneficial aspects of a full complementary circuit, Ron noted, is that if any output transistor ever should short out, it will instantly take its opposite-polarity companion with it, so that under no circumstances can the full dc power supply voltage of either polarity ever appear on the speaker line. In some other amplifier designs, as sad experience has shown, a transistor failure can send a speaker up in smoke and flames by feeding the full dc supply to it.

When asked about TIM distortion, Ron generally agreed with Bob Carver that the majority of amplifiers on the market probably have a sufficiently rapid slew rate that TIM won't be a serious factor. However, he suggested that one reason for audible differences among amplifiers may be a behavior known as "conditional instability," in which if the amplifier is driven into clipping or into slew-rate-limiting (especially with highly reactive load impedances), it starts generating ultrasonic oscillation whose audible side effect may be the "glassy" quality some critics hear in some amps. (For those who are curious, the slew rate of the Dreadnaught amps is 25 volts per microsecond.)

Asked about the new VFET designs, Ron noted that their high cost is due not only to the high development cost, which must be recouped from profits, but also because the several output devices in each circuit must be carefully hand-matched for electrical characteristics, a factor which may also lead to high repair costs if the burnout of one VFET requires the replacement of the entire set. Further the VFET's use a high bias voltage and operate more nearly in class A than in the usual class B, which means that the amps will absorb a lot of electrical current, contain a heavy and costly power supply, and run hot.

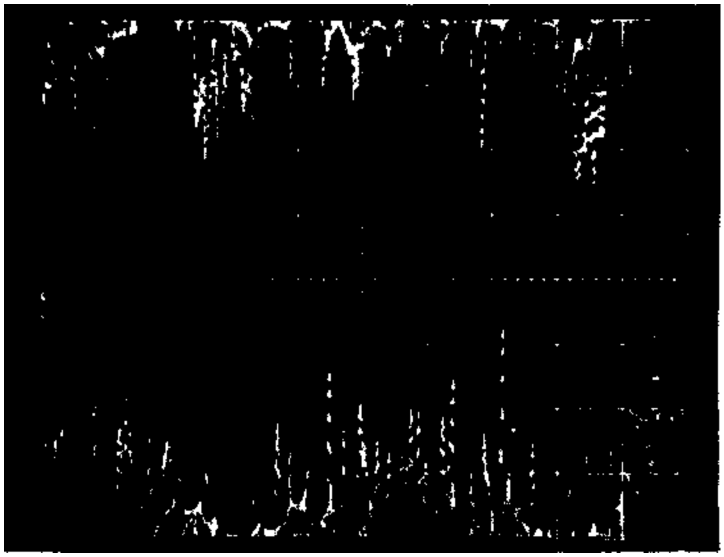
Two amplifiers which Dunlap did mention favorably in passing during the course of his talk were the current BGW units, which can successfully drive low-impedance current-hungry loads, and the classic AR integrated amp, which has sufficient current-output capability that it really can deliver 90 watts into 4-ohm AR speakers.

The second half of the meeting consisted of an extended demonstration of the peak currents which real loudspeakers actually demand in reproducing music at high levels. A dual-trace oscilloscope showed the output voltage from the amp on one trace and the current being delivered to the load displayed on the second trace, as measured by a Hall-effect current probe clamped on the wire going to the speaker (see the photograph and explanation on the next page). Dynamic passages of music were used, notably selections from the DG recording to Bizet's "Carmen" at the Met and the climax of Ravel's "La Valse" in an Ozawa/BSO broadcast which had been Victorized (i.e., broadcast without compression or peak-limiting).

With an AR LST it was observed that midrange material (Marilyn Horne's voice) produced large peak voltages at moderate currents while bass-drum impacts caused large current peaks. At integrated sound pressure levels of about 100 dB SPL, peak currents as high as 10 amps were noted, though the indicated voltages were not very high, indicating a relatively low effective impedance. With a Smaller Advent generating maximum SPL readings of about 105 dB, peak currents of 12 to 14 amps and peak voltages of 80 volts (the limit of the amplifier) were seen, indicating an effective impedance not quite as low as the LST, and in this case the highest current peaks were associated with cymbal crashes in the music. Incidentally, these high current peaks were observed despite the presence of a 3-amp AGC fuse in the speaker line to protect the speaker from excess currents lasting longer than a few milliseconds.

An EPI 50, played at about 100 dB SPL, drew 7-amp peaks mainly at low frequencies. Finally, two loudspeakers Made by a member evidently had a true 8-ohm impedance, as it was possible to drive the amplifier into peak voltage clipping (at 105 to 110 dB SPL) with maximum current drains of only about 7 amps.

Not surprisingly, many members left the meeting both with a clearer understanding of power amplifiers and with a new concern about the difficulties that can arise at the amplifier/loudspeaker interface. If nothing else, the demonstrations were an effective reminder of the basic equation that defines power as the product of both voltage and current, and plainly the success of a power amplifier depends on its abilities (and limitations) in both of these areas.—Peter Mitchell



Loudspeakers are not resistors. This photo of an oscilloscope display shows that loudspeakers are dynamic impedances, not simple resistances. The upper trace shows how voltage varies at the loudspeaker terminals in response to a music signal, in this case a bit of the finale from "La Valse." The lower trace displays the current in the speaker lines. Note that the two traces are almost completely different. They show what the amplifier is supplying to the speaker at any given instant, the horizontal axis being time.

If loudspeakers were purely resistive, the current and voltage traces would peak simultaneously—they would point at each other. But because loudspeakers are sometimes inductive and sometimes capacitive reactances, depending on frequency, and because current and voltage "lead" or "lag" each other, depending on the nature of the reactance, these traces reflect sometimes nearly all current or nearly all voltage, but rarely simultaneous voltage and current peaks.

This makes amplifier testing into resistors unrealistic. No resistor will treat your amplifier's protection circuits or output transistors nearly so harshly as a loudspeaker.

This may help explain why some amplifiers seem to make some speakers sound better than do others—it may be less a matter of wattage (although that is important) than of intelligent design, which takes the complex impedance characteristics of loudspeakers into account.

The traces show the "positive-going" waveforms in each case (the upper trace was inverted for this photo). Vertical sensitivity was adjusted for clear display rather than to show clipping or other malfunction. Voltage trace sensitivity was 5.0 volts/cm, and current trace sensitivity was 500 mA/cm. The loudspeaker was a Smaller Advent yielding peak SPL's of about 90 dB measured at a distance of 3 feet on axis. This translates to a relatively low level for home listening, and makes a good case for amplifiers with high voltage and current capabilities.—Jim Brinton

Making a Compact Headphone Amplifier (or Two)

Peter W. Mitchell

PART ONE: A MINIATURE POWER AMP

In recent years I have preferred the Sennheiser 414 headphone because of its combination of good sound and minimal wearing discomfort. So when I developed a need for a headphone amp for monitoring on-location recordings, I put together a little battery-powered unit employing type 741 op amp integrated circuits (described in the September 1974 Speaker). It drove the 2000-ohm impedance of the Sennheisers very nicely. However, as is well known, the 741 and similar op amp IC's are essentially voltage multipliers which work best into load impedances of 1000 ohms or higher; they were not designed to drive low-impedance loads such as 8-ohm headphones.

So when I recently discovered how dramatically superior in sound the Koss Pro-4AA is, * it became necessary to find or develop a little amplifier in order to be able to use the Pro-4AA for on-location monitoring. The selection criteria were: good quality, minimum size and weight, minimum cost, and ease of construction. The Southwest Technical Products headphone amp looks inviting, but is costs $45 and occupies 200 cubic inches of space, critical when I am trying to fit a complete recording system into two suitcases. Another possibility is to get the schematic of a known good headphone amp (such as the HP stage in the Advent 202-HP) and build a copy of it; but such a circuit involves 40 or 50 parts and thus fails the ease-of-construction requirement.

Devoted as I am to sloth, I like integrated circuits. And nearly every IC manufacturer makes power-amp IC's, designed for use in TV sets and auto radios, rated to deliver from 0.5 to 5 watts into an 8-ohm load. Unfortunately most of them are quite unsuitable for hi-fi use, exhibiting either elevated distortion (1% or more, plus crossover distortion) or a rapid rolloff in response below

*Like most people I had previously compared headphones only in stores. There, with phones being driven from an unfamiliar amp with unfamiliar source material in a noisy environment, one could tell that the Koss Pro-4AA sounded different from the Sennheiser, but no more. When I recently had an opportunity to make an extended at-home comparison among four phones (Koss Pro-4AA and HV-1LC and Sennheiser 414 and 424), it became possible to define the differences among them. The Pro-4AA is by far the most accurate of the four; each of the others sounds quite colored in comparison, though each is pleasant to listen to and is much more comfortable than the Pro-4AA. Of course there are some electrostatics that are even better, but in my view they are not enough better to justify their cost. The headphone amplifiers described in this article are not designed to drive electrostatic phones effectively.

about 100 Hz. An exception to this trend is the National LM380, whose response is flat from sub- to ultra-sonic and whose distortion is specified as under 0.2% across the entire audio band at output levels up to 4 volts rms (2 watts into 8 ohms). Furthermore its specified distortion remains low at lower output levels, indicating an absence of crossover distortion. As a final bonus, the 380 requires fewer outboard components to get it working than any other power-amp IC that I've seen, so construction is relatively simple.

Circuit Description

The schematic of one channel of the 380 headphone amp is shown in Fig. 1. The gain of the IC is 34 dB, too high for convenient use, so an input volume control is necessary unless you will use the amp only with signal sources having output level controls. For monitoring on-location recordings it is useful to try to obtain a dual level control whose two channels track accurately, so that you can correctly judge the stereo balance of a recording as you set the mike levels.

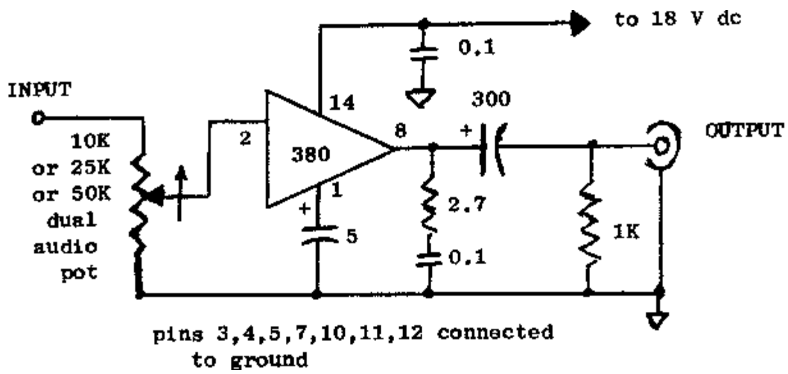


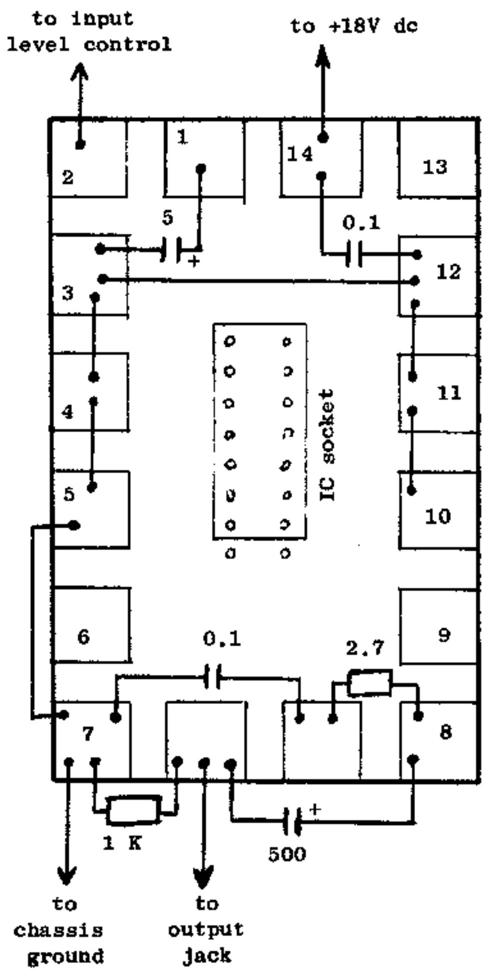
Fig. 1. Schematic of LM380 headphone amp (capacitances in microfarads)

The schematic indicates the IC pin numbers for the 14-pin DIP format. The 5-µ F electro-lytic capacitor connected from pin 1 to ground forms part of a power supply decoupling filter. The input signal goes to pin 2 and the IC output appears at pin 8. A 0.1-µ F disc capacitor is con-nected from pin 14 to ground to guard against RF interference on the power supply line. Pins 3, 4, 5, 7, 10, 11, and 12 are all connected to ground. The IC is capable of delivering about 2 watts of power to an 8-ohm loudspeaker load, but in that application pins 3, 4, 5, 10, 11, and 12 must be connected to a heat sink, making the construction task substantially more complex. The heat sink can safely be omitted if the IC will be used to drive only dynamic headphones.

Nearly all modern solid-state power amplifiers require a filter at the output to suppress ultrasonic oscillation; the 2.7-ohm resistor and 0.1-µF disc capacitor serve that purpose here. The signal then passes to the output jack through an electrolytic capacitor which blocks the 9 volts dc residing at pin 8. This capacitor, together with the impedance of the headphone, comprise a filter which rolls off low frequencies; so the value of the output capacitor should be selected with the expected headphone impedance (Z) and the desired low-end limit in mind. The frequency at which the response will be 3 dB down is f = 160,000/CZ, where C is in microfarads. In the proto-type of the amp I used 300 µF; if you will use only phones of 250 ohms or higher impedance, you can reduce C to 50 F. In any case, the output capacitor should have a rated working voltage of 10 to 16 volts. Finally, at the output end of the capacitor, a resistor is used to carry off any leakage current that may accumulate when the headphones are not plugged in; the value of this resistor may be anything from a few hundred to a few thousand ohms.

Construction Hints

Each channel of the headphone amp can conveniently be constructed on Radio Shack's IC "socket adapter" printed-circuit board. Fig. 2 shows the suggested layout. Note that the PC board has 16 holes for the IC socket, while a 14-pin IC socket is used for this IC. By placing the socket in the top 14 holes, the circuit-board pads corresponding to the unused bottom two holes can be utilized for the output circuitry. The IC pin numbers are indicated in the corresponding pads in Fig. 2.



PARTS LIST

Qty.			Radio Shank
2	LM380N ICs	$4.00	276-1725
2	14 pin sockets	1.20	276-1999
2	PC boards	1.00	276-024
2	5 µF 16V	.80	272-1001
4	0.1 µF	1.60	272-1069
2	500 pF 16V	2.00	272-1007
2	1000 ohms	.25	271-000
2	phono jacks	4/1.50	274-346
1	**headphone jack**	2/1.70	274-312

plus 2 2.7 ohm resistors, a chassis box (LMB 138 or equivalent), and a DC power supply.

Fig. 2. Layout and parts list

The LM380 will operate on any dc voltage from +10 to +20 volts. Its current drain (for two channels) is about 12 mA in the absence of any audio signal, rising to 15 or 20 mA at average signal levels and to peaks of about 50 mA when driving Koss Pro-4AA's to 110 dB SPL. The peak current drain may be even higher if headphones are used that are at once low in impedance and poor in sensitivity. Therefore small 9-volt transistor radio batteries connected in series to obtain 18 volts cannot be used. If battery operation is required, it would be necessary to make up a battery pack with 8 to 12 flashlight batteries (C or D cells) in series, or use large lantern batteries. Of course, if you are recording where you can plug your recorder into ac, you can also plug in an ac power supply for the headphone amp; the Lafayette 99F50742 or Olson BA-133, for about $10, probably would be suitable. If there is sufficient demand from members, a

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regulated power supply could be designed specifically for the 380 amp. The most convenient approach of all, of course, is to tap into your recorder's power to drive the headphone amp; the practicality of this depends on the design of your recorder and upon your willingness to risk voiding your machine's warranty. In the case of the Advent 201, for example, a simple internal modification enables the existing 18-volt jack to be used to power the headphone amp. However, because of inadequate power-supply decoupling, the Advent mike preamp cannot simultaneously be driven from the same jack and would have to be run from a series-connected pair of 9-volt batteries.

PART TWO: THE 741 STRIKES AGAIN

The type 741 op amp IC (see "The Audiophile's Friend," September 1974 Speaker) was designed as a small-signal amplifier, not as a power amp. It is generally considered to be unsuitable for driving load impedances below 1000 ohms; indeed the standard spec sheet for the 741 includes a graph showing that its "output voltage swing" (the maximum peak-to-peak signal outout) falls off rapidly as the load impedance is reduced below 1000 ohms. Now it is generally true in audio that if you try to make a device drive a lower impedance than it was designed for, the signal level goes down and the distortion goes up. So while the 741 drives 2000-ohm Sennheisers nicely, it would be expected to distort badly if you forced it to drive the Koss Pro-4AA or other low-impedance phones.

However, when Howard Souther (designer of the Pro-4AA) appeared on "Shop Talk" recently he revealed that the true impedance of the Pro-4AA is not 8 ohms but 250. And a followup investigation shows that, indeed, many so-called 4- to 15-ohm phones actually have impedances of 100 ohms or more. So one may reasonably wonder: is it definite that the 741 cannot drive the Pro-4AA and other headphones? Might it be at least marginally usable (and thus attractive when using a tape recorder which has no headphone output at all)?

To find out, I connected the output of a 741 amp stage to a distortion analyzer, drove the 741 to an output signal level of 3 volts rms, and varied the load impedance, expecting to see the distortion rise continuously as the impedance went down. Surprisingly the THD remained at a constant low level until the impedance was reduced to 300 ohms, at which point the 741 went into clipping, refusing to drive that impedance to that high a signal level. When the signal level was set at 1 volt rms, the THD remained low until the impedance was reduced to below 100 ohms, and with a signal level of 0.5 volt, the impedance could be reduced to less than 50 ohms without the amp distorting. Fig. 3 illustrates these results. Evidently the 741 can indeed deliver low-distortion signals to a relatively low-impedance load, though it will do so only at a reduced signal level.

To clarify this situation I measured the variation of the 741's output clipping level versus load impedance, and the result is shown as the "voltage" curve in Fig. 4. With a 2000-ohm load, the 741 will deliver a signal of up to 4.8 volts rms at low distortion, while with an 8-ohm load, the distortion remains low only up to 0.1 volt rms. The graph also shows the current delivered to the load; as the load impedance is reduced and the maximum undistorted signal voltage falls, the output current rapidly rises until it reaches 20 mA, the limit permitted by the short-circuit protection in the 741. Consequently the most striking of the three curves in Fig. 4 is the one showing the maximum undistorted power output that the 741 will deliver into each load impedance. The maximum power (voltage x current) varies considerably with impedance, and by a delightful coincidence the 741 will deliver the most power into an impedance of about 250 ohms. This seems to be a generally unrecognized characteristic of the 741 op amp, but is a very handy one indeed.

Is a maximum power of 35 milliwatts (0.035 watt) enough? To those of us accustomed to 350-watt amplifiers it doesn't seem like much, but then a headphone has to fill only a few cubic inches of air with sound, not a 3000-cubic-foot room. As it turns out, when fed 35 mW, the Koss Pro-4AA

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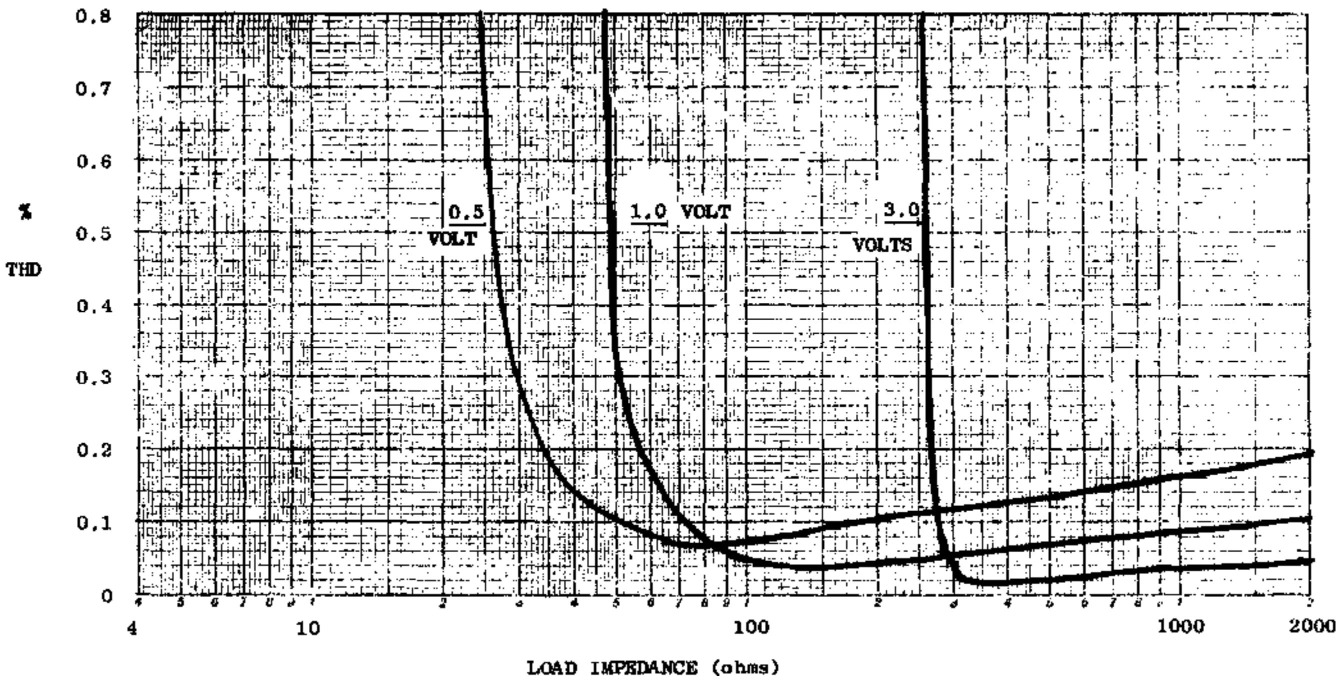

Fig. 3. THD (including noise) versus load impedance at 400 Hz
(type 741 op amp IC, ±9-volt supply)

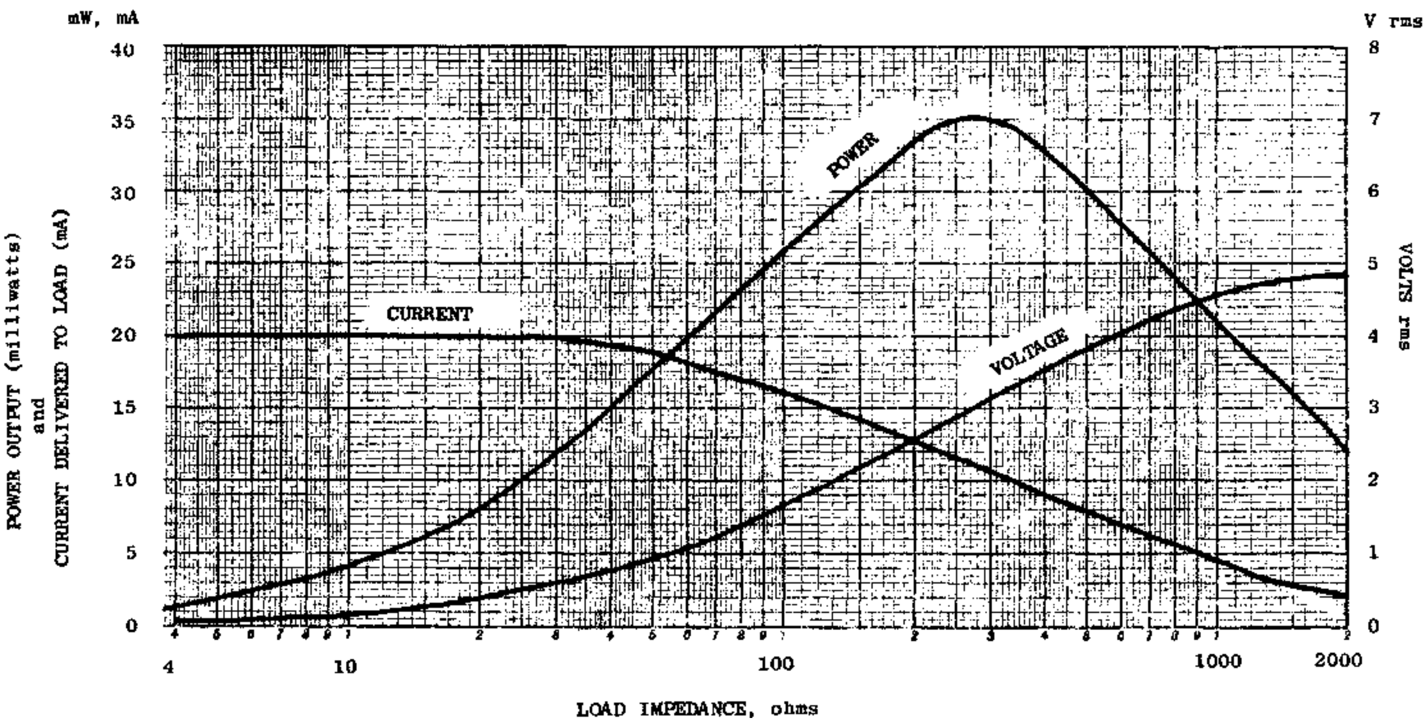

Fig. 4. Continuous power output of type 741 op amp IC at 400 Hz
(power supply ±9-volt batteries)

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produces a sound level of 110 dB SPL, an ample level for on-location monitoring and most other listening as well. Of course, with music having a reasonably large peak-to-average ratio, one actually would adopt a maximum sustained listening level of 95 to 100 dB SPL in order to avoid clipping off the peaks—and to preserve one's hearing.

So far so good, but what about a headphone whose impedance really is low? Fortunately there tends to be a correlation between impedance and sensitivity in dynamic headphones: units having a true 8-ohm impedance often are much more sensitive than the Pro-4AA, so the 741 can drive them fairly well, too. The following table lists a few examples, based on manufacturer's specs; unfortunately, credible impedance and sensitivity specs are unavailable for many headphones.

Headphone	Impedance	Maximum 741 Output, mW	Sensitivity (Power Required for 100 dB), mW	Maximum SPL with 741, dB
Koss Pro-4AA	250	35	3.5	110
Koss HV-1LC	150	30	4.0	105
Marantz SD5	8	3	2.8	100
Pickering OA3	15	6	0.6	110
Pioneer 405	8	3	0.6	107
Scintrex 98	300	35	6.5	107
Sennheiser 414	2000	12	0.8	112
Sennheiser 424	2000	12	1.0	111

The schematic of the basic 741 headphone amp is shown in Fig, 5. If you will use the unit only with signal sources that have their own output level controls, then you can use this fixed-gain version of the headphone amp. The 270K feedback resistor provides 15 dB of gain. If you want less gain, you can decrease the feedback resistor in each channel; for example, 150K will give 10 dB.

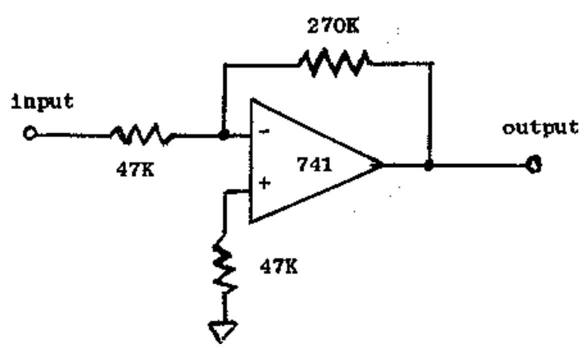

Fig. 5. Schematic of basic 741 headphone amp

Fig. 6 shows a simple variable-gain version, using a dual pot for the feedback resistance in each channel. Use this version with signal sources lacking output level controls. To permit correct judgment of stereo balance in setting recording levels, the two channels of the 50K pot must track accurately. If you can't find a suitable pot, you could use a 2-pole, 6-position switch to make a level control that operates in discrete steps of about 3 dB. Pairs of resistors of 5% tolerance will ensure that the two channels of the amp are matched within 1/2 dB. Fig. 7 shows the circuit for each channel. The switch must be a "shorting" (make-before-break) type, such as the Calectro E2-164 or NAE EB-164, but not the Radio Shack 275-1386.

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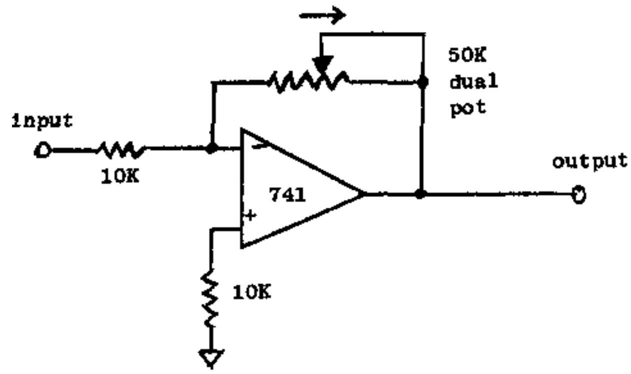

Fig. 6. Variable gain version of 741 headphone amp

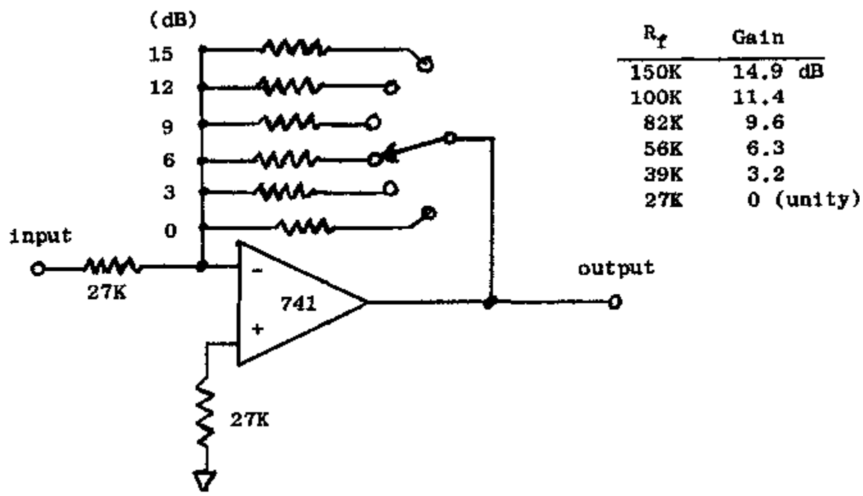

Fig. 7. Variable gain version using step level control

Speaking of switches, one that is very handy to have in any headphone amp is an input switch to enable one to monitor left only, right only, or stereo. The best-buy switch for this use is the Lafayette 99-61566, which at 80 cents costs about half the usual price for such switches. It is a 4-pole, 3-position switch, but of course only two of the poles are used here. The wiring recommended for the input switch (as seen from the bottom) is shown in Fig. 8.

<u>Construction</u>

The 741 op amp is available in several formats. For example, you can use the 5558V (dual 741) mini-DIP, the 747 full-size dual 741 in a 14-pin DIP, or two single 741 8-pin mini-DIP's installed in a 16-pin socket. In any case the circuit is easily built on the Radio Shack 276-024 "socket-adapter" PC board, whose copper-clad pads can be drilled with a no. 60 bit for installing the resistors and wires. Obviously the circuit board layout will depend on the IC format that you choose. If there is a sufficient demand from members, a kit of parts can be supplied for about $15, with the circuit board pre-drilled and with full wiring instructions.

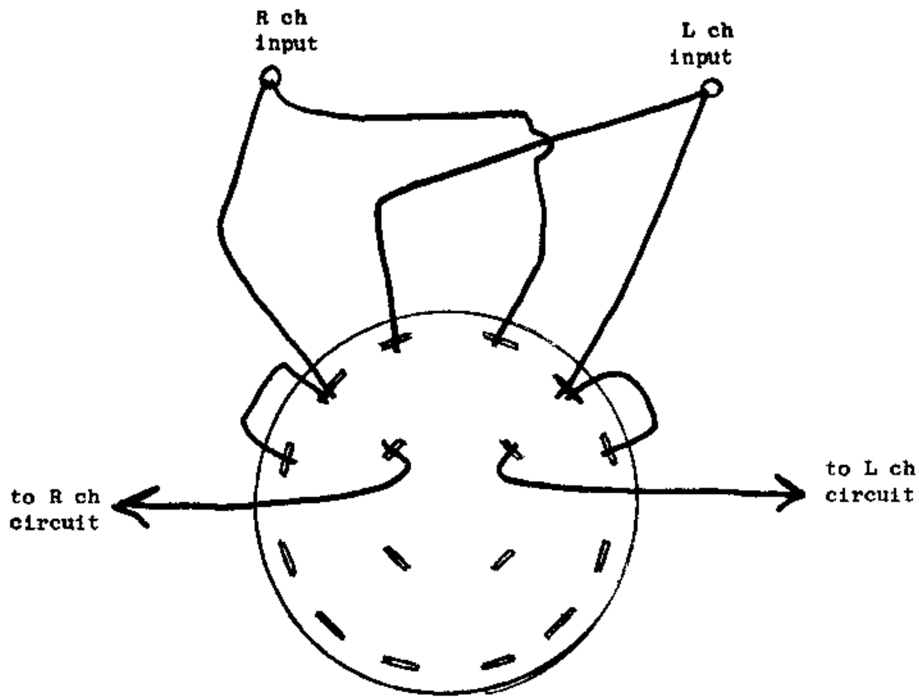

Fig. 8. Input switch wiring (Lafayette 99-61566)

The op amp operates on matched positive and negative power supplies. Using Pro-4AA headphones, the current drain from each polarity is about 7 mA at average signal levels, rising to 10 or 15 mA on loud peaks. So ordinary 9-volt transistor radio batteries will power the circuit for at least 10 hours of use.

PART THREE: WHICH IS BETTER?

Since two very different headphone amplifiers have been described, we are faced with a problem of choice. Some differences already are evident: the 741 headphone amp is somewhat cheaper and easier to build than the 380, and is much better suited to battery operation. On the other hand, the 741's power output drops off dramatically at low impedances, so it will drive low-Z headphones only if they are very sensitive, whereas the 380 will deliver its maximum power into low impedances. Specifically, at load impedances of 250 ohms or higher, the 741 and the 380 will deliver nearly the same maximum power, but at impedances below 250 ohms, the 741's power-output curve falls rapidly to 3 mW at 8 ohms while the 380's maximum-power curve continues rising to about 2000 mW at 8 ohms (but only with proper heat-sinking of the IC). Clearly, if you are using low-Z phones which are insensitive, the 380 is the better bet. Incidentally, if you don't know the true impedance of your dynamic headphones, a very good approximation can be obtained by measuring the dc voice-coil resistance using a good ohmmeter.

With headphones of 100 ohms or higher, the 741 is generally the better bet. It not only is easier to build and use, it also sounds a little better. On the majority of musical material the two amps sound identical. But on mixed choruses and on some other material when you are monitoring the direct feed from the mikes (rather than listening to a recording which already has some distortion in it), the 741 is audibly clearer and more transparent than the 380. This difference,

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incidentally, is much more obvious with Pro-4AA's than with Sennheiser 414's. The 741 also is quieter than the 380, though this difference too is audible only in the live mike feed; the 380's noise is masked by tape hiss when playing back recordings, and by surface noise and preamp hiss when listening to records. Finally, the 741 measures better than the 380. The following measurements (except for the clipping level and corresponding maximum SPL) were made with each amp driving a 250-ohm load to a 1-volt level, which corresponds to 100 dB SPL with the Koss Pro-4AA.

	380	741
Output at clipping	3.3 V	2.9 V
Maximum SPL (Pro-4AA)	111 dB	110 dB
Frequency response (20 Hz to 20 kHz)	Flat	Flat
THD (including noise)	0.15%	0.05%
IM distortion	0.32%	0.01%
Unweighted S/N	68 dB	85 dB

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Audio Myths

Daniel Shanefield

Here are some statements deeply believed by many audiophiles, amateurs and professionals alike. In a marathon sequence of experiments over the past few months, I have found to my amazement that they are mostly hollow myths.

If a researcher is going to attack such things as these in print, he really ought to give more detailed descriptions of the experiments than I have, but I just don't have the time. So rather than keep silent, I will try in this article to summarize the experiments concisely, but with enough description so that knowledgeable readers can check my results. (Probably only barely enough description, although confirmation by others would be important here.)

Please duplicate some of my experiments. I'll bet very few of you will believe my contentions without seeing all this on your own oscilloscopes, so well-entrenched are these myths in our hi-fi psyches.

The Myths

1) "Corner placement of loudspeakers extends deep bass." Who ever perpetrated this in the first place? As anyone can see from the response curves in Roy Allison's papers (see the April 1974 Speaker, page 6, and the Journal of the Audio Engineering Society, Vol. 22, No. 5, 1974), corner placement has no significant effect on the low-frequency rolloff, that is, on the frequency at which the knee in the curve appears. I easily verified this with a sine-wave generator and a factory-calibrated AKG 202E mike driving a Hewlett-Packard 400FL ac voltmeter. Corner placement does provide an illusion of deep bass, by intensifying various mid-bass peaks and by increasing the integrated efficiency of the whole bass range. But you could increase the latter just as well by turning up the bass tone control knob, and you wouldn't get such aggravated peaks and valleys.

Allison's speaker placement ideas (see his Fig. 15D in the JAES article) worked well in my particular listening room. For example, good results were obtained with speakers placed on the floor, against the side walls, about 4 feet from one end wall, and aimed toward the other end wall. Of course one should avoid distances that are submultiples of room dimensions. There still will be peaks and valleys in the response curve, but not as many or as bad as with corner placement.

2) "A high damping factor tends to give a tight bass sound." Who promulgated this prevarication? In a double-blind test with a witch-or-glitch switch, I put a 16-ohm resistor in series with a loudspeaker in one channel, and then turned the gain up a hair to compensate for this. Quick comparisons were made to the other channel speaker directly without a resistor. Monaural music sources were used through a good system, which included a Dyna 400 power amp and short, thick

wires. I couldn't hear any difference, even when using a "crossoverless" speaker (a Bose 901 turned around to face me). More conventional speakers such as the EPI 100 and the Dyna A-25 were also used, with the same results. Other good amplifiers with damping factors from 50 up were also tried. Results were negative each time.

Puzzled, I ran a wide variety of tone bursts through the systems, using a sine-wave generator and the normally closed contacts of a simple relay buzzer. The mike was a Sony ECM 280 condenser, going through an HP 400FL VTVM's internal amplifier (checked by also using a Nakamichi tape recorder's transistorized mike amp), terminating at an oscilloscope. The 16-ohm (and other) resistors gave no noticeable degradation of the damping at any frequency from 30 Hz to 19 kHz, even at the resonant frequencies of the speakers.

I thought, "It's really quite simple; I'm just making some kind of dumb mistake." I certainly didn't have the nerve to try telling anyone that "The Emperor has no clothes." But then, in an exchange of taped conversations with Julian Hirsch, he mentioned that damping factor is baloney. Emboldened, I re-repeated some of the experiments, placing the speakers up on a ladder out in the back yard, to get a nearly anechoic response. No difference.

It might seem possible that the damping factor would have an effect on an otherwise poorly damped, low-quality speaker. But I couldn't hear or see any such effect on an old Lafayette speaker (model number unknown) or a new Radio Shack Solo 103.

Right now I'm ready to say that bragging about high damping factor is useless, and maybe a little bit dishonest.

3) "<u>Output transformers inevitably degrade fidelity</u>." Absolute inanity. I put the 8-ohm terminals of an old Williamson-type transformer across the left channel output from my power amplifier. The 16-ohm terminals of the transformer were attached to a loudspeaker. (One terminal was common, thus making use of the "autotransformer" principle, in which only the secondary windings are connected.) The right channel's signal path went through an 8-ohm <u>resistor</u> instead, and then through a witch-or-glitch switch which could allow that right-hand signal to drive the same speaker. I adjusted the left channel of my graphic equalizer to maintain a flat frequency response. Result: no difference in sound. There was some measurable phase distortion, but it was <u>not audible</u> in a double-blind test. (I wish I had a dollar for every inductor in the signal path at the recording studio, mixdown board, cutter system, etc., or for every degree of phase distortion in the studio's matching amps, bridging amps, 70-volt line-driver amps, etc.) Transformers might be hard to design, when one is considering today's high standards of response flatness, but they are not inherently bad.

4) "<u>Complex musical waveforms cause problems such as TIM, etc.</u>" Baloney, spumoni, and biscuit tartoni! Attach the vertical input of your oscilloscope to the input of your preamp or power amp, and the horizontal input to the amp output. With equal input amplitudes and modern transistorized hi-fi equipment, you'll always see a 45° straight line, unless you've got phase distortion.

If you have an experimental setup that does demonstrate TIM, just try inducing it with a square wave from a disc recording, instead of from a function generator. The slew rates of cutter heads, pickup styli, etc., are simply too slow to cause TIM, tone control ringing, and several other sophisticated-sounding but actually mythical problems in modern hi-fi equipment.

After all this, is everything rosy, after all? Well, live-versus-recorded tests indicate that the answer is potentially "Yes." All you have to do is debunk the myths and concentrate on what are really the most important problems.

The weakest link in a good system is usually the recording, as others have said before. But enough good recordings exist so that we can say it is feasible to record realistically.

The next weakest link is usually the loudspeaker. My tests (and tastes) tend to favor the Magnapan MG-11 (formerly called the MG-2167), even over the big Magnaplanars.

In most cases one also needs synthesized (or quadraphonic) reverberation from one or two speakers at the rear. Also, I recommend some degree of reflection from the front walls; a bipolar speaker like the MG-11 standing about four feet from the front is excellent in this respect.

My own experiments in trying to provide a large-hall feeling in a small listening room support the Ambiphon company's claim that very low frequency sounds are useful. (See page 15 of the April BAS Speaker, and listen to the narrative of the Ambiphon demo tape called "The Sound of Space.") That does not seem to be a myth, although I must say that Ambiphon's commercially available tapes don't seem to be any better in this respect than some other good recordings. Very realistic hall sounds can be heard in the Acoustic Research's demo record ENY-AR-1. Another example, and a rather extreme one, is band 7, side 1, of the Stereo Review binaural demo record.

Unfortunately, when you try to hear these deep bass sounds on loudspeakers, you will probably bump into what seems to be another audio myth, or at least a semi-myth. And that is the concept of 20 Hz to 20 kHz sound. Very few speakers emit much of the "space-creating" deep bass.

There are some exceptions, though. To beef up my bass-shy Magnapans, I use the Bose 901, facing forward and mounted in authentic Roy Allison style. This provides an active area equivalent to that of a single-cone speaker of 13.5-inch diameter, plus a huge amount of electromagnetic coupling. It rolls off with a knee at 30 Hz, which is as far down as I can hear anyhow. I've biamped it through an active crossover, to limit the response to the range of from 30 to about 100 Hz. At these low frequencies there don't seem to be any audible distortion problems in the 901.

From double-blind tests of live-versus-recorded music, it appears that the other components of the system need be of only middle quality, provided a graphic equalizer is used to straighten things out. Anything measurably better than that is usually not audibly better.

One last recommendation: squash the devil out of the mid-bass region, using the equalizer. Go far beyond what pink noise tests would tell you to do. It's another semi-myth that pink noise is adequate for equalizing a room. It's not repetitive for a long enough time to allow resonances to build up fully, but real music *is*. The same goes for warble tones: they're inadequately repetitive. You often want to catch those slowly built-up resonances and compensate for them, not ignore them.

A brief but practical procedure is to use pink noise first, and then decrease the mid-bass response further, until the speakers don't sound like boxes any more. (Even Maggies and electrostatics can sound boxy in a living room.) Before you get used to it, this new type of sound might seem "thin," as some people also say initially about the sound of the Philips RH 532 feedback system. (See Stereo Review, March 1975, page 44.) But it is more realistic.

After going through all this, I still can't say whether my system duplicates the concert hall sound, because I can't quickly A-B the two of them. But my philosophical muse says, "So what?" And she's just as much of an audiophiliac as I am. (That's someone who has audiophilia to a pathologically compulsive degree.) She says, "The studios do various synthetic things to the music anyhow, like adding presence peaks, adding reverb, adding 'snare drum sound' to the drums with a Kepex, etc. This actually makes it sound better in some cases—sort of like putting you up closer. As long as the reproduction has now gotten so good that you can't say you're sure it's worse than the original, who cares ?" There are a few parts of a few records where I agree with her.

In place of these myths, I therefore suggest the following points of my own. Concentrate on (1) super recordings, (2) aggressive equalization, (3) super speakers, (4) some very deep bass, (5) some added room reflections, and (6) some tough defensive logic, when you sense that the sound is getting good.

Phase Distortion and Transient Response

Dennis Colin

I anticipate that the mere mention of these two overused terms is conducive to interest and skepticism, and dangerous to your music listening pleasure. In fact, they have nearly destroyed mine, for I have been using my high-fidelity components largely for research into phase distortion, transient response, and other possible excuses for reproduced sounds being different from original ones.

My research has been extremely rewarding to me, and although not performed with the rigors of statistical analysis, has yielded results that can be agreed upon by different people in different environments.

Importance of Phase and Transient Response

Transient response refers to the accuracy with which a signal of short duration is reproduced. Transients contain a wide, continuous distribution of frequencies. The perfect example is the impulse (a very short, but intense pulse), which, if perfect, would contain every frequency from dc to infinity.

For audio purposes, it must contain at least the whole audio range. Fourier and LaPlace found that if you added together all frequencies at equal amplitude and in phase, the result would be zero everywhere except at one time: an impulse.

So, two things can mess up transient response: variations in either amplitude versus frequency response or phase versus frequency response.

Most transients in music are the attacks of instruments, but not all. Even the "steady-state" cycles of a violin or trumpet tone, for example, are always changing, and have transient noises such as scratches or puffs of air.

It is these transients that provide the pinpoint directional accuracy of the ear, as well as some timbre identification. Have you ever felt that your system was giving a great illusion, only to instantaneously focus your head on the exact location of a buzzing fly.

Electronically Generated Effects

My first experiment was to generate an electrical impulse using a synthesizer. It was a pulse of 10-microsecond duration and 1-second repetition rate. When played through my Sennheiser HD-414 phones, it sounded like a click. Through an AR-2aX it was similar, except for (1) slightly different coloration, (2) room echoes, and (3) a noticeable frequency dispersion, i.e., it sounded "chirpy," the high frequencies being heard before the lows.* I then filtered the pulse to produce a "Gaussian" pulse, i.e., with rolled-off highs and lows, and adjustable peak frequency, but all frequencies still in phase. I found the "chirping" worst near the crossover frequencies of the speaker. Later, I looked at the impulse response of the speaker using a Thermo Electron 814C mike, whose own impulse response I found to be very accurate, using an electric spark. Sure enough, the speaker does delay lower frequencies more.

Then I connected up a specially designed network that electronically delays lower frequencies more than high ones, in a gradual manner. The effect was very noticeable, and similar to the speaker's own effect.

I found, incidentally, no effect on continuous periodic tones, such as square waves, above about 1 kHz, On recordings of percussion, bass, and acoustic guitar, the electronic phase distortion was noticeable, seeming to destroy "punch" and clarity.

Incidentally, the AR-2aX seemed to have no worse phase distortion than any other dynamic speaker system I have tested. Even one full-range Bose 901 driver by itself had some. Actually, this had less "chirp" effect, but unfortunately, the Bose driver had more high-end coloration and directivity. With the AR-2aX, however, there were phase and amplitude distortions due to driver interference, especially at off-axis angles. Listening with two ears tends to mask them, since both ears will hardly ever hear the same effect.

One further very interesting experiment was to synthesize a tone of about 16 Hz where the higher harmonics came first then the lower ones, and a second tone identical in frequencies but backwards, i.e., lower harmonics first. The first one (tone A) sounds like a downward rapidly repeating chirp, while the second one (tone B) is an upward chirp. The strange thing is that tone A always seems to come from behind the speaker, while tone B seems to be in front of it! Tone A is similar to an impulse that has the low frequencies delayed more than the highs, which is what almost every speaker, tape recorder, and equalizer does.

The opposite distortion, where high frequencies are delayed more, is very complex to do electronically, and almost never occurs in components. I do not know why delaying lows more than highs makes the sound distant, but only that it does. Could this be why recordings sound buried compared to the original?

Experiments With Real Sounds

Asking myself this, I went to a friendly recording studio and made a master tape on an Ampex 16-track recorder using one Neumann U-67 omnidirectional microphone. I recorded drums, cymbals, acoustic guitar, piano, and electric bass (the latter plugged in directly—no microphone). I then transferred this tape onto a half-track stereo tape using an Ampex machine. On one track, I played the master and recorded the copy in the normal forward direction. On the other track, I played the master backwards, and turned the copy backwards, too. The result was two copied tracks side by side, both forwards, but one copied backwards.

Theoretically, the backwards process will cancel any time or phase distortion due to different delay of different frequencies (if the two machines have similar head characteristics) while the forward copying will double it.

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*On Shop Talk last winter, Mark Davis discussed phase shift effects and demonstrated the audibility of phase shift on impulses by playing back pure pulses alternated with phase-shifted pulses that sounded "chirpy." Cassette recordings of this broadcast are available from the BAS for $2.00 plus cassette.

I have since played the tape to at least four other people, on four different speaker systems: AR-2aX and AR-4 (acoustic suspension), JBL L-100 (bass reflex), and Altec 604E (reflex and horn), and on a pair of Sennheiser phones. We all agree that the backwards copy sounds better. It is difficult to describe the difference further, except to say that the backwards one is cleaner, more natural, and less buried. Interestingly, the difference is not just in the attack transients; the sustained parts of the guitar notes were also apparently affected.

I have recently tried two more experiments. In the first, I shorted out the series inductor of my AR-2aX. This made a slight difference in frequency response, as noticed by listening to pink noise. (Better or worse I'm not sure.) But it does improve the transient response, both on music and oscilloscope impulse response.

The effect was not due to elimination of the time delay. Removing the inductor and its resistance lowered the Q of the woofer system, which in turn improved its transient response, but at the cost of rolling off the deep bass.

In the second experiment, I made up a speaker system consisting of a Bose 501 10-inch woofer and a Bose 901 4 1/2-inch full-range driver for the highs. There is considerable overlap in response, so I needed only a capacitor in series with the small driver. The system does not have extended highs (manufacturer's data for the 901 driver shows rapid falloff above 13 kHz). However, the transient response, to drums anyway, is very good; but the significant thing here is that it sounds best when the tweeter is a few inches behind the woofer, to compensate for the ever-present low-frequency lag of big cones and high-inductance voice coils.