Volume 1, Number 5 Winter 1977/78

The Audio Critic®

In this issue:

The speaker survey continues, with glad tidings in the medium-priced sector and mostly unhappy discoveries in high-end esoterica.

We have our first look at headphones, including some rather impressive electrostatics.

Our power amplifier survey comes to its epilogue, but not before a new superstar is born.

We evaluate a whole new generation of preamps and find exciting evidence of progress.

We hail a quantum jump in cartridge performance and have good news in arms and turntables as well.

Plus the expected and unexpected extras.

The Audio Critic

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The Audio Critic is an advisory service and technical review for consumers of high-priced audio equipment. It is published at intervals averaging two to three months by The Audio Critic, Inc., and is available by subscription only. To maintain total dedication to the consumer's point of view, **The Audio Critic** carries no advertising by equipment manufacturers, distributors, reps, dealers or other commercial interests. Any conclusion, rating, recommendation, criticism or caveat published by **The Audio Critic** represents the personal findings and judgments of the Editor and the Staff, based only on the equipment available to their scrutiny and on their knowledge of the subject, and is therefore not offered to the reader as an infallible truth nor as an irreversible opinion applying to all extant and forthcoming samples of a particular product. Address all editorial correspondence to The Editor, The Audio Critic, Box 392, Bronxville, New York 10708.

For subscription information and rates, see inside back cover.

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All Right, We Promise: No More Promises (A New Policy Statement from the Publisher)

As you can see by looking out of the window, this issue isn' coming to you in December, nor in January for that matter. It took a great deal longer than we had anticipated.

If you found the wait frustrating, you may be in no mood to recognize that even so the interval since the last issue has been shorter than the shortest interval between issues of any other noncommercial audio publication. You may also be unwilling to give us credit for having published our first five issues within a span of a little over 14½ months, an unheard-of frequency among such publications. What you probably have in mind instead is why the hell we don't publish on a bimonthly schedule, as originally promised. The answer to that is in our editorial in this issue, which we beg you to read.

As we explain there, our perspective has changed. We're currently publishing information unavailable from any other source, and we find it very difficult to schedule in advance the solution of unsolved problems, which is what a lot of this information is about. Consequently, we've decided to stop making any promises regarding our mailing dates. The new policy is: you get it when you get it.

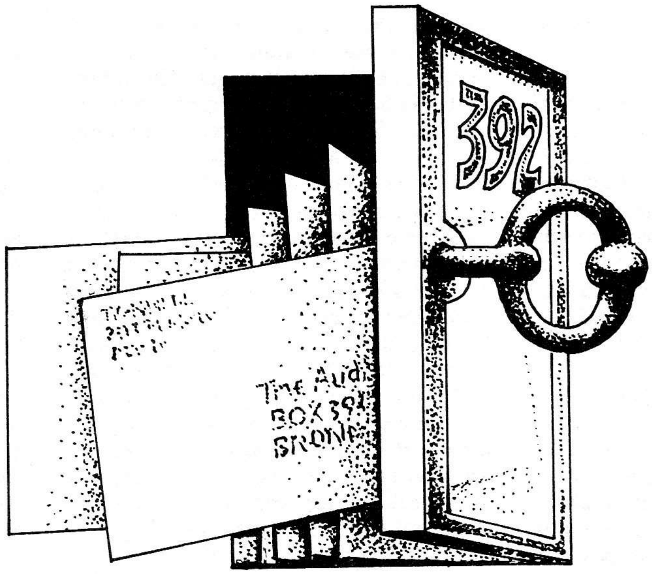
That doesn't mean we aren't planning to publish with greater frequency in the future than in the past. We're still hoping to become a true bimonthly—eventually. Until then we anticipate a long-term average interval of two to three months between issues, which is one reason why we aren't changing our six-issue subscription policy (the other is bookkeeping). Remember now, this is an estimate rather than a promise. With a little bit of bad luck, such as the paralyzing snowfalls, flu epidemics and equipment failures of our past winter, we could have another long lapse between issues.

One immediate corollary of our new policy is that we don't answer letters about the nondelivery and nonreceipt of issues that haven't been published yet. Please make a mental note of this once and for all, as we've run out of stamps and secretarial time for such unnecessary correspondence. If the issue is out, it will reach a correct address by first-class mail with almost 100% reliability; if the issue isn't out yet, no amount of letter writing can expedite it by even one day.

We thank you for your patience, understanding and continued support.

Box 392

Letters to the Editor



For the first time, in addition to subscriber correspondence from both the professional and the private sector, we're including reactions and rebuttals from manufacturers and equipment designers whose products we have reviewed in past issues. Our criterion for printing such material is the correspondent's attempt at reasoned and factual argument, whether the alleged facts are true or not. Purely vituperative and uninformative who-the-hell-do-you-think-you-are letters have been, and will continue to be, ignored. The letters we publish in this column may or may not be excerpted, at the discretion of the Editor. Ellipsis (. . .) indicates omission. Address all editorial correspondence to The Editor, The Audio Critic, Box 392, Bronxville, New York 10708.

First, the letters of general editorial interest, starting with a real mind-stretcher from the world's smartest audio designer below the legal age of adulthood.

The Audio Critic:

I can sympathize with your frustrations over the search for conclusive amplifier testing. Trying to quantify those nebulous characteristics which we call "musical accuracy" can often seem impossible. I think, though, that partly responsible for the repeated failures encountered by critics and designers alike is a desire for one test which will tell everything one needs to know about a system's performance without requiring any other test data, knowledge of circuit design, or basic understanding of the nature of a musical signal.

Take, for example, your "pet" test, CCIF IMD. This test had been yielding some reasonably correlatable results until suddenly you encountered the condition where two amplifiers measured drastically differently, but sounded almost equally good. The assumption here should not be that the test is invalid (after all, as Chris Russell of Bryston stated in your issue Number 4, "distortion is distortion"), but rather that Amplifier One, which is not at all fazed by this test, makes some other error, which Amplifier Two carefully avoids. The net audible effect, call it the total distortion of each, is approximately the same.

Now you are faced with the situation where you know that there are at least *two* conditions which must be tested for, but you don't even know what the second one is. When I reached this juncture some time ago, I decided to forget all of the preconceived notions and attitudes which led me to this predicament, and begin from the beginning. I felt that a better knowledge of the kinds of errors occurring in amplifiers, as well as an understanding of how and why they exist, would lead to a more conclusive and correlatable body of testing procedures.

Any sound can be characterized by a change in amplitude over some period of time. This can be expressed as $\Delta A/\Delta t$ or, in the limit as Δt approaches zero, as dA/dt, where A is the amplitude and t is time. This is all Edison needed to know in order to invent the phonograph, and it's all we need to know to examine the performance of an amplifier. Any attempts to transform this basic truth into the frequency domain, or any other set of coordinates, assumes conditions (i.e., periodicity, etc.) which serve only to complicate matters. Remember that while harmonic structure can be represented as amplitude versus time, the reverse is not necessarily true. Frequency domain test methods have become our standard as their results are quite easy to observe. It's too bad that generally they don't correlate with what we hear.

Expressing sound as dA/dt, we can characterize the complete function of any amplifier as "output equals gain times input," or, for input equal to dA/dt, outputs equals $G(dA/dt)$, where G equals the gain of the amplifier. This leads to two, and only two, possible families of distortions:

The first type of distortion is a result of variability of gain. For an amplifier to pass a signal undistorted, its instantaneous gain (that is, the gain at any point in time) must be absolutely constant. Harmonic distortion is simply the case where gain varies with amplitude, usually due to nonlinearities in device transfer functions. In a completely distortion-free amplifier it would be possible to divide any output signal into a series of vanishingly small time integrals and find that the gain of the amplifier for each of these periods is the same.

The second type of distortion is caused by a change in time base characteristics. From the above expression of any amplifier's function, it is clear that there should be no effect whatsoever on the time component of the input signal. That is, for any interval 'dt' in the input signal its value must be unchanged by the amplifier. For example, if the input were composed of two pulses separated by, say, one millisecond, they should appear at the output exactly one millisecond apart. If their width at input is, say, one hundred microseconds then their width at output must also be one hundred microseconds. Unfortunately, in real amplifiers things are not this simple.

Take this case of a signal composed of two pulses. We know that real amplifiers will not respond to this signal instantaneously, but will exhibit some propagation delay. If this delay is the same for both pulses, then the distance between them will be unchanged from input to output. However, if the instantaneous forward delay of the amplifier is not absolutely constant, and one pulse is delayed longer than the other, then the distance between them will not be accurately preserved. Thus, in a distortion-free amplifier, time delay must be totally invariant.

Now that we have established the two basic performance criteria for all amplifiers, we can examine what happens to signals in real circuits. Take, for instance, the "emitter follower", a very common transistor power amplification configuration. Because of the effects of junction capacitance there will be some delay inherent to this circuit. This delay is a function of emitter current which is, in turn, partly a function of base voltage. As this current is increased the capability to overcome the junction capacitance is greater, thus reducing the delay. The problem is that increased signal amplitude results in higher base voltages, and the result is obvious: this is one where propagation delay is not constant, but varies with amplitude.

This leads to an interesting discussion of the Class A phenomenon. In general, Class A stages tend to sound better than Class AB stages. This difference has been commonly ascribed to the absence of any form of crossover notch in Class A configurations, yet none of the better Class AB amplifiers exhibit any such aberration. (This distortion is easily measured as increased THD or IMD with decreased signal amplitude.) The reason that Class A is capable of sounding better, in some cases, than Class AB is simply that the output stages are constantly handling such high currents that they are easily able to overcome delay producing capacitances. Because the values of the delays are shorter, any resulting changes in delays are shorter, and thus less audible. Given that increased current always results in increased amplitude-related distortions, as well as decreased time distortions, the fact that Class A can sound better than AB indicates that maybe time is more important than amplitude. Of course, understanding these ideas should make it possible to design an AB configuration with minimum time *and* amplitude distortions.

Heat is another factor which can result in variable propagation delays. The laws of physics tell us that conductivity is directly proportional to heat; as the junction temperature of a transistor increases it becomes a better conductor, reducing its effective capacitance. This means that as an amplifier heats up, its delay time is reduced. Most amplifiers, especially those designed to operate "cool to the touch", exhibit rapid changes in junction temperature and thus rapid changes in delay times. Consider the case of two high-amplitude pulses appearing at the input of a cool amplifier. The first pulse is delayed in reaching its peak value, but once it has, the resultant increased current demands on the circuit cause a sudden increase in operating temperature. By the time the second pulse reaches the amplifier, the devices are no longer cool, and the delay time has changed. Here again Class A designs excel because not only are the values of delay changes smaller, but the enormous thermal capacity afforded by their necessarily mammoth heat sinks slows down temperature shifts.

All of this leads to another interesting point. It has been generally accepted that amplifiers with capabilities for very wide bandwidth sound better than those whose bandwidth is limited, even though we can't hear above twenty kilohertz or below twenty hertz. The fact is that every amplifier can be considered a bandpass filter, and depending on the exact characteristics of the equivalent network, this filter can have quite variable delays. Those amplifiers incorporating a 12 dB per octave high pass filter, whose rolloff begins just below twenty hertz, exhibit changes in delay times a decade or more above the cutoff point. Thus delay in these amplifiers will change with frequency. Extending the passband of a circuit above and below audibility will push these time nonlinearities out of the audible range.

In addition, one cannot overlook the effects which reactive loads have on amplifiers. Since present circuits are far from ideal voltage sources (they all have some source impedance and reactance), their interaction with loudspeaker networks results in some filter circuits having definite time characteristics. As the source impedance of the amplifier is decreased, the effects of this phenomenon will become minimized. Also, for reactive loads, the current demands placed on the amplifier in order to maintain a given output voltage can change, thus possibly changing the delay characteristics of the circuit.

It is tempting, at this point, to assume that if the delay of an amplifier is constant it can otherwise be ignored. Unfortunately, with most amplifiers this is not the case. As long as feedback is involved, delay of any kind is an important matter. During its forward propagation delay period an amplifier is operating open loop, resulting in a short burst of increased "instantaneous gain". As defined by our original representation of an amplifier's function, this is distortion. Fortunately, most amplifiers are fast enough so that this simple aberration is below the threshold of audibility. Unfortunately, in most amplifiers the condition is prolonged by overload. When the open loop amplifier clips, it can often take many times its propagation delay to once again become linear. This phenomenon can be quite audible. Susceptibility of an amplifier to this condition is a function of its delay time, amount of feedback used, overload recovery characteristics, and signal rise time.

Simply put, to prevent the amplifier from overloading, the product of propagation delay times signal slew rate should never exceed the maximum output of the amplifier divided by its open loop gain. If this condition is met, then there will be some feedback returned to the amplifier input before the signal reaches a level which will clip the open loop amplifier. Ensuring that this is the case involves minimizing propagation delay, minimizing open loop gain (and thus feedback), increasing maximum output capability and/or filtering (slowing down) the input signal. Therefore, in order to evaluate the effects of a given propagation delay for a specific amplifier one must know something about its circuit design. That is, if two amplifiers have the same constant propagation delay, the one with less feedback, greater headroom, better overload recovery, and a filtered input will probably sound better. However, if one has a lot of feedback, but good overload recovery, and the other has no headroom, but a filtered input . . .

These are the kinds of problems which cause audible distortions. Until our testing procedures are designed to discover such inaccuracies we will never be able to separate better from best on the laboratory bench.

Very truly yours,
Andrew S. Rappaport
A. S. Rappaport Company, Inc.
Armonk, NY

Cf. our comments introducing Part III of the power amplifier survey in this issue. (Thank you, Andy, for some exceedingly nourishing food for thought.)

—Ed.

The Audio Critic:

. . . Concerning the use of separate woofers/subwoofers: If the time domain is so important, and pulse response measurement becomes likewise important in loudspeaker measurement, why doesn't the use of a physically displaced woofer cause poor time domain response and poor pulse response measurement? Unless I misunderstand what I have read on the subject, a pulse, because it is of no specific frequency, acts as all frequencies at once; therefore, part of the pulse would be reproduced by the woofer. If this were so, if the woofer were displaced, as with the DQ-1W behind the DQ-10's or in conjunction with the Rogers, wouldn't the pulse be distorted? (Or even with the Beveridge 2SW using the subwoofers.) Evidently this is not the case, as borne out by measurement and listening tests (or else you would have not failed to mention it), but I am unable to figure out why. Could you please elucidate? . . .

Sincerely,
Michael Steiner
APO San Francisco

You're absolutely right; a separate subwoofer can be time-aligned with respect to the main system only with one specific placement, if at all, which is unlikely to be the placement used in an actual listening situation. Luckily the ear is increasingly less sensitive to wave-front coherence as we go lower in frequency; at typical subwoofer crossover frequencies (say, 70 to 120 Hz) it's quite insensitive. In our pulse tests, we use pulse widths of 1 msec to 0.1 msec, corresponding to half-wavelengths of 500 Hz to 5 kHz. That's where the action is when it comes to time coherence.

—Ed.

The Audio Critic:

I would like to have an accurate speaker system. But I have one big problem (along with most other audiophiles, I imagine): I cannot afford anything that fits in that category.

You and your "consulting engineers and other technical advisers" seem to know what it takes to make a reasonably accurate speaker system. Would it be possible for you and/or your fellow worker(s) to design a system that would pass the scrutiny of your panel of golden ears and would not be too expensive; then write an article on where to get the drivers and how to build the system?

If this is at possible, I and a lot of other stereo nuts would be very grateful. If not, well I tried and thank you very much anyway for your time.

Sincerely,
J. Dennis Smith
Belle, WV

A reference loudspeaker made of conventional (meaning nonproprietary) drivers is definitely one of our future projects. Whether the typical audiophile will be able to duplicate it without the aid of a calibrated microphone, signal generators, oscilloscope, spectrum analyzer, etc., remains to be seen. Don't hold your breath, though, until this project comes to fruition.

—Ed.

The Audio Critic:

Having plowed through the Mark Davis correspondence, I think that Mr. Davis has a point. This is not to say that I completely agree with him, but he has done the homework and run controlled tests, and you have not. The point, in other words, is that he has established the minimum conditions which must be met in order to test the significance of any parameters other than frequency response. Therefore, if you run comparative tests without satisfying those conditions first, your conclusions are as questionable as if you twisted your cartridge around at random between A-B tests.

If, as you say, you are serious about trying to correlate measurements with sound, and if you find Mr. Davis' conclusions unacceptable, then the *only* way to refute him is to duplicate his tests using your equipment and see if he is right. Informal tests and "the painstakingly accumulated wisdom of an entire generation of audio perfectionists" just won't do it—after all, for hundreds of years the accumulated wisdom was that the earth was flat. And, Mark Davis notes in his letter that in several *informal* tests he was able to hear differences which *disappeared upon formal testing*. To simply state that his system had insufficient resolving power without showing that a better system would demonstrate differences is giving in to the "golden ear mysticism" that you deplore. And trying to find more subtle objective-subjective correlations without first eliminating the known effects of frequency response and level variations is simply sloppy.

Having said all this, I do want to say that I think you have presented a lot of good, solid information in your publication and I hope your efforts to find "good" measurements succeed. It is simply common sense, as well as good science, that the more sensitive the measurement, the more careful one has to be to eliminate any possible extraneous factors.

Sincerely,
James Lin
St. Paul, MN

You're quite right on a general, philosophical basis, but you're ignoring the specifics of the situation. When a renowned practitioner announces a startling and controversial discovery, it does indeed behoove other practitioners to spend some time and effort on duplicating his experiments. You forget, however, that Mark Davis isn't exactly a Maxwell or a Helmholtz. He's the guy who blew his credibility when he denied the existence of TIM and impugned the research of Matti Otala (and distinguished predecessors). So if Mark Davis should suddenly announce that smearing chicken fat on the transistors improves transient response, we wouldn't necessarily drop everything then and there and run to the deli to initiate a verification. On the other hand, we might publish his letter on the subject because it would probably amuse and stimulate our readers.

Quite independently of the above, however, the new preamplifier survey reported in this issue did include listening comparisons between units with exactly matched frequency response and signal level—not because of Mark Davis but because it was easy and convenient to do so. And, what do you know, we heard distinct differences. Read it and weep.

—Ed.

The Audio Critic:

I object to the philosophy in Mark Davis's letter that the pursuit of excellence in high fidelity is a waste of time if it goes beyond the known psychoacoustic limits of the ear. The goal of high fidelity is to reconstruct a time-varying sound pressure field that spatially and temporarily duplicates some original field. The ear is irrelevant to the *statement* of the goal. The goal is objective, not subjective. It is a matter of physics and not of psychoacoustics.

Psychoacousticians can provide guidance whenever a design trade-off must be made, to make the best choice. But when no trade-off is required, when the SOTA permits 0.0008% distortion equally well as 0.2% distortion, who is Mark Davis to say I'm foolish to prefer 0.0008%? Likewise with TIM. If TIM exists and we can eliminate it, then let's do so. Who cares whether it's audible?

I am glad to see that someone besides Richard Heyser understands the importance of a loudspeaker's impulse response and will defend it in print. It is a fact of mathematics that if a loudspeaker operating in its linear range has a good impulse response, then it must have a good frequency response (flat magnitude response and linear phase response). The converse is also true. Good frequency response (magnitude and phase) implies good impulse response. However, a good magnitude response alone does not imply a good impulse response.

There is no excuse for any loudspeaker designer today not to use impulse response testing . . .

Sincerely,
Stephen D. Stearns
Research & Development Engineer
GTE Sylvania
Mountain View, CA

The Audio Critic:

In your most recent "Admonitor" you censure Epicure and Bose for sophistry in the design of their speakers. While I agree with your conclusions regarding the generation of time delay in the two speaker systems, the argument you presented was simplistic. The reverberation recorded during a performance is inadequate for completely portraying the acoustics of the concert hall. In fact, recording engineers deliberately attempt to exclude much of the natural reverberations of the hall by using close-mike and directional-mike recording techniques. The reverberation is added subsequently using artificial means. Excluding the natural reverberation is actually desirable. If, as you state, the full character of the reverberant sound were retained in a recording, then ideal recordings could be obtained by placing two microphones at your favorite listening position spaced in correspondence to the distance between your ears. Unfortunately, the result of recording using this attractively simple technique is that the sound becomes garbled. The admixing of the echoes results from the loss of vital information in the process of recording—information about the directionality of the individual echoes making up the reverberant sound. Without the directional information all the echoes are homogenized, producing an aesthetically displeasing result. Theoretically two channels of recorded sound should be sufficient to convey the required information. After all, our two ears detect only two signals in the original experience. The problem is recording the proper signals. Stereophonic recording is fundamentally incapable of accurately conveying directionality, particularly when the virutal source is not located between the two speakers. Binaural recording must account for directionality in the reverberant sound field, information which is not conveyed in the present art.

A fundamentally different approach is to synthesize the required information. Instead of attempting to capture the echoes of the hall in the original recording, the delayed replicas are generated during the playback process according to our knowledge of the situation in which the recording was made. Unfortunately, this technique has practical problems as well. A typical concert hall has thousands of significant echoes. The early echoes are especially important in determining the acoustical character of the hall. To effectively recreate the sensation of the original hall, the timing, amplitude, and direction of each echo need to be carefully controlled. The problem with the two speakers which you justifiably criticized is not that they attempt to generate the echoes, but simply that the temporal and directional pattern of the echoes which they do generate is wrong. The echo response of the speakers is dependent upon the geometry of the home listening room. Accordingly, the directionality of the echoes will correspond to the listening room and not the concert hall. Furthermore, the delays will be about an order of magnitude smaller than in a concert hall—milliseconds instead of tens of milliseconds. Your ear will not resolve two signals spaced by only a few milliseconds. The result will be what you described as blurring or smearing of the sound. Nevertheless, this distortion is an implementational problem which is not inherent in the synthesization approach. Synthesizing the reverberation during playback is basically a valid and powerful approach.

One of the most important problems still remaining with high-fidelity reproduction is accurately portraying the ambience of the original hall. Traditional recording techniques do not realistically convey reverberation because they discard directional information. Synthesizing the reverberation during playback is a valid approach but the echoes must be generated properly or, as in so many things, the cure will be worse than the disease.

Very truly yours,
Jeffrey Borish
Sound Technology, Inc.
Campbell, CA

We find this very sad. Here's a maker of sophisticated instrumentation for measuring audio accuracy (we have a Sound Technology distortion measurement system in our own laboratory), and what is he doing? He is arguing on behalf of inaccurate audio!

The best way to dispose of all arguments in favor of "synthesized" reverberation or any other tampering with the signal is to listen to Volume One of the Mark Levinson Acoustic Recordings (an organ and choral program) on a properly designed and aligned playback system. The record was made with two microphones straight into the tape recorder, tape recorder straight into the cutter amplifier, no signal processing, no added reverb, no black boxes, no nothing. And the acoustics of Dwight Memorial Chapel at Yale are there *beyond the wall of your listening room, big as life. End of discussion—next!*

(But see also Max Wilcox's article on the same subject in the back of this issue.)

—Ed.

The Audio Critic:

A simple misreading of the definition of β, and no "careless trig," caused me to erroneously measure L sin β of the Grace G-707 and SME tonearms, and then to contradict your claim that "tonearm designers (have) forgotten their high school geometry."

(We agree. Your misreading was indeed simple.—Ed.)

I don't quite understand the intensity of the editorial blast that resulted *(Have you forgotten your remarks about our "irresponsible sloppiness," causing our "credibility to deteriorate rapidly," etc., etc.?—Ed.)*, but it is quite clear that, having correctly found one error in one of my arguments (I apologize profusely; consider yourself lucky that it didn't cost you $28 to read it) *(huh?—Ed.)*, you then implied that all my remarks are "just plain wrong" and proceeded to "make an example of them." Unfortunately, you sidestepped the larger technical questions I was attempting to bring to your attention:

1. What good is correct tonearm geometry if you can't *exactly* align your cartridge with that arm? This question is mainly directed to tonearm and cartridge manufacturers, who could quite easily produce self-aligning cartridges and headshells, along with the correct geometry (I hope) they are now designing as a result of your article. Until such a design is available, we are left with the tedious and inaccurate alignment method described in your issue #4.

There is a very interesting sentence contained in the description of that method (p. 55): "Then you twist the cartridge in the headshell so that you can see absolutely no tracking error at those two points." Please explain to us how, without benefit of a transparent headshell, this can be accomplished with the accuracy of 0.1 degrees which you imply is necessary? Up until that point, the alignment procedures are quite sound, but it is at the crucial point of angular alignment of the cartridge that all of this hard-won accuracy is thrown away.

(Not thrown away, but—you're right—in jeopardy. This is by far the most troublesome, least predictable and therefore least formalizable part of the alignment. With cartridge bodies having all right angles, as in the Denon DL-103 series, it's relatively easy. With more complicated shapes, such as the GAS 'Sleeping Beauty' series, it can be a real pain. A small mirror, accurately scored with the appropriate guidelines, is one way around the problem. Our next issue—Number 6—will contain additional information and suggestions on this very subject. Meanwhile, we can assure you that doing the alignment as best you can, minor inaccuracies and all, will still get you a whole order of magnitude better performance than not doing it at all.—Ed.)

2. At no place in my letters did I suggest or intend to suggest that a Bessel (Q = .58) response was either optimum or practical for a woofer. As anyone who reads my letters can see, I was merely correcting your obviously erroneous contention that Butterworth tuning provides the best possible transient response.

(Butterworth tuning does provide the best possible transient response while maintaining flat amplitude response to the lowest possible frequency. —Ed.)

In a larger context, it needs to be made clear that Butterworth and Bessel responses are merely mathematical abstractions: two particular points on an infinite continuum (a multi-dimensional continuum, by the way) of filter tunings; that Butterworth is a calculated set of conditions which (it can be proved) provides the maximum bandwidth of a system with completely monotonic (i.e., no ripples) frequency-amplitude response; and that Bessel provides a roughly analogous frequency-delay response. The whole point of my remark was that neither you nor I nor anyone else has ever shown that any particular tuning point on this continuum of possibilities is either "correct" or "the best tradeoff between amplitude and time response."

(This is sheer sophistry. Do you mean, for example, that in a sealed-box woofer the "correctness" of Q = 2.5 is equally defensible—or its "incorrectness" unprovable? Come on, Alan. You've proved to the fans that you, too, know something about filter theory; now be practical.—Ed.)

To make matters worse, in the reviews of loudspeakers which follow, you repeatedly chant the litany of the importance of time domain over frequency domain, and go so far as to review very highly a speaker (Cizek Model #1) which provides a switch position for either Q = 1 or Q = .6. Q = .6 is awfully close to Q = .58 (Bessel). You go on to describe the bass as "close to state of the art . . . solid, well-defined, musical," after having replied to me that this response has "pretty grim amplitude characteristics." In addition, you maintain that for this speaker "the damping is absolutely correct," and then immediately describe the two dampings that are available at the flick of a switch. Are they *both* "absolutely correct"?

(Yup. In a world full of sealed-box woofers with a Q of 1.7 and 2 and even 2.5, either 0.6 or 1 is damn near correct. "Absolutely" may not have been the best-chosen word, but you know very well that you're just hassling us for having caught you with your trig down.—Ed.)

3. The RCA Philadelphia/Ormandy recordings (by and large) sound bad. Max Wilcox was the producer of those recordings. The producer of a recording is responsible for its sound quality. Therefore, Max Wilcox is responsible for a very large number of bad sounding recordings of a very fine orchestra. It doesn't particularly matter to me if he recorded them in the seventh floor men's room. They sound bad. Why? Didn't he have any influence over the choice of recording site? Is the recording site *all* that is wrong with the RCA Philadelphia/Ormandy recordings?

(Well, you see Alan, Max really wanted to fly the orchestra to the Concertgebouw in Amsterdam, but those crass skinflints at RCA said no.—Ed.)

Audio Critic #4 raises some similar questions: on page 35 you make the statement that the Q of the Ohm F is roughly 1.4, unless it is placed in the middle of the room, "in which case the Q is approximately correct." ("Correct"? Here we go again . . .) The Q of the loudspeaker changes with room placement? I think it is about time you described to your readers (for your benefit as well as theirs) just exactly what "Q" means.

(You're a bit confused here. The Ohm F is one of the very few speakers that can be realistically analyzed as looking into a 4 π (full-sphere) space; the "correctness" of Q = 0.707 is based on a 2 π (half-sphere) space. The "correctness" of Q = 1.414 for a 4 π space is therefore at least defensible.—Ed.)

Regarding your use of nearfield measurement, if you will refer to Don B. Keele's paper on the subject, you will see that a fundamental premise of that technique is that the speaker is a direct radiating rigid flat circular piston, etc., etc., and *that an Ohm F most certainly is not! (You see?—Ed.)* The main idea of nearfield loudspeaker measurement is to allow one to measure the loudspeaker's "anechoic" response in a non-anechoic environment. If you can change the measured response of a loudspeaker by relocating it in your room, your measurements are clearly not independent of the room, and therefore not successfully emulating anechoic conditions.

Impulse testing, in the full sense of the word (which means quite a bit more than just looking at pulses on a scope) reveals both time and frequency domain characteristics (which are merely different views of the same thing) rather completely. Impulse testing is in no way *opposed* to amplitude response measurement, as you seem to imply (issue 3, p. 8, issue 4, p. 28). In particular, the "near perfect response" you refer to on p. 8 of issue 3 implies not only minimum time smear, but it also indicates near perfect amplitude response. The fact that the DCM speaker shows "ragged" frequency response indicates that its impulse response isn't really "near perfect"; only that it *looks* that way on a scope. Don't forget that the ear is capable of detecting probably one part per million errors, while the eye at best can see one part per *hundred* errors on an oscilloscope display. The real power of impulse testing is that the resulting device output can be analyzed to provide extensive and precise time *and* frequency domain information. Journalistic imprecision with the concept of impulse testing could provoke a frequency response vs. impulse response civil war that would be just as confused as its tube vs. transistor, TIM vs. feedback, and belt drive vs. direct drive predecessors; but with a lot less technical basis. I think your readers would enjoy a careful discussion of the time vs. frequency "problem," as well as the related measurements that are now available (3-D plots, FFT, etc.) and that this could well prevent the kind of confusion referred to above.

(Here you're absolutely right—it's the second Tuesday of the month, so it's your turn—and we're indeed planning all sorts of goodies on the subject.—Ed.)

When I wrote out my check for $28 I had very high hopes for your magazine. Since that time those hopes have largely been fulfilled, and I have been quite gratified by your accurate and concise reviews, refreshingly broad and profound insight into audio, and your relatively high technical competence (who else talks about Thiele's and Small's work, for instance?). Perhaps I am being too demanding in asking you to refrain from participating in the name-drop and technical mysticism cult, the new-buzz-word-a-month club, which seems to be the core of the audio scene. It seems that everyone, before he can be spoken to or about, must be labelled as a mathematical type or academician, expert or amateur, golden-ears or techno-freak. Complicating this phenomenon are the members of the Harry Pearson/James Bongiorno school of thought, who seem to believe that technical ignorance can be disguised with so much invective and name-calling.

The published critic puts himself in a very dangerous position, vis-a-vis the clearly productive artist or engineer, when he is intemperate in his (however valid) criticism of the other person's creative labors (for example, your often petty remarks inserted in Mark Davis' letters). I humbly suggest that you refrain from this kind of sniping; it is really quite irrelevant to your avowed purpose, as well as (I believe) the interests of most of your readers. Furthermore, it casts a pall of small-mindedness and egotism over what is otherwise quite high quality work. I, and most other audiophiles I know, have the greatest respect for your efforts and the resulting magazine; please don't endanger the credibility of those efforts by participating in the petty paranoia and oversimplified technical horn-blowing which seems so rampant among audiophiles. I think most of your readers are more interested in the qualities of the equipment you review than offhand remarks about the perceived competence of other audiophiles, equipment designers, and in some cases, your own subscribers. It seems particularly a shame to dilute reader interest and confidence in this way, in light of the high quality of your equipment reviews and technical articles.

Sincerely yours,
Alan S. Watkins
Burroughs Corporation
Pasadena, CA

Well, well, look who's preaching temperance all of a sudden . . . We happen to disagree with you, even though your basic perceptions are quite in line with ours. We feel, however, that when a charlatan launches a new product, for example, it isn't enough to tell our subscribers that the product doesn't sound (and measure) as it should. That's much too abstract. It's extremely useful for the consumer to understand that the man is a charlatan. High-end audio is by and large a one-man-one-design kind of field, and the credibility of the designer (or maker) is actually a more important issue than the performance of a single product. We want our subscribers to understand that from certain practitioners they can expect clear thinking and sophisticated product design, and from others just the opposite. In other words, we want our subscribers to know exactly what we know, i.e., what we would tell our personal friends if we didn't have a publication.

—Ed.

* * *

And now a few words from our reviewees.

The Audio Critic:

Considering the entire report on the Dual CS 721, we would probably be well advised to forego any comment. However, our appreciation of your customary precision and evidence of fair play prompts us to risk whatever editorial comment may in turn result. Thus emboldened, we have one quibble and two somewhat more serious observations to make.

The quibble is with your use of the word "except" between the statement that "We don't find anything seriously wrong with it, except that (a certain manual combination) is even better and costs $60 less." Perhaps a new sentence beginning with "However" might have been a bit more felicitous. After all, no matter how many other choices may exist, they don't make anything seriously wrong with the CS 721.

(That's a matter of perspective. It could be argued that there's something seriously wrong with a given choice when a better choice is easily available. But why quibble with a quibble?—Ed.)

Later, you express concern about the mechanism that locks the cartridge holder to the headshell because of the *"possibility"* (italics ours) under *"worst-case conditions"* (italics also ours) that there may be "play (and) lack of positional repeatability." Certainly, those end results, if they existed, would disturb a tracking-error fanatic, as you state.

Since there is a locking three-point fixed position, with no play possible, we don't quite understand where you found any lack of positive positioning. (Incidentally, this entire design was awarded patent number 3,247,032, a copy of which is attached.)

(*Well, we found it was possible to lock the cartridge holder into several positions that differed by a hair either horizontally or vertically. It isn't a precision device—nor is a patent a guarantee of quality. Of course, we're talking about very small variances, but then it takes a shift of less than 0.2 degrees in some cases to change the sound of a record grove.—Ed.*)

Our other point has to do with the anti-resonant mechanical filters, which you do acknowledge with heady praise as not a "totally wrong-headed solution," given certain designs and price considerations. Since this device resulted from considerable Research & Development and produces demonstrable and repeatable results, we'd be interested in knowing what tests might have been made with balsa wood or silicone gunk that make them "definitely more desirable." Certainly they would be innovative, even if not commercially practical.

(Come on, Murray, you want additional R & D for $28?—Ed.)

However, all things considered, we have no quarrel with your final conclusion that the CS 721 might be "best for the money if you must have an automatic."

Sincerely yours,
Murray I. Rosenberg
General Manager
United Audio
Mount Vernon, NY

The following response to our review of the RAM 512 power amplifier, by Mr. RAM himself, was preceded 11 days earlier by a shorter letter signed "Peter Ledermann, Electronic Engineer, RAM Audio Systems, Inc.", which in our opinion fell into the mindlessly-hostile-and-uninformative category and is therefore not allowed a free soapbox here.

The Audio Critic:

In reference to your article, I would like to discuss the facts omitted, sidetracked, distorted and overbiased in your review, Vol. 1, No. 4.

1. I never told you or suggested that I was the M of C/M Laboratories, on the contrary, I told you that I was *not* the M, but was chief design engineer of their high-fidelity products. I was, however, a founding partner of Audio International, Inc./C/M Laboratories in Connecticut.

(You never told us anything at all on the subject, but who cares?—Ed.)

2. The RAM 512 is indeed an all-out design in the 180/180 watt (8 ohms) power class. We at RAM Audio Systems are dedicated to one thing and one thing only, and that is the limited production of the highest accuracy audio reproduction electronics. We follow the basic industry standards for pricing as based on raw materials, parts cost, assembly labor cost, warranty cost, sales and distribution cost, and retail dealer profit; if that is "exorbitant" then the entire electronics industry pricing structure is wrong. Remember Arrow Electronics' "we will be here when you need us"?

3. The RAM 512 is probably the strongest mechanically and best-finished power amplifier in the world. *(The world? —Ed.)* The interior subchassis is formed from three (3) closed 'C' channel 18-gauge (0.049 in. thick) cold-rolled steel structures, and the rear panel that is 11-gauge (0.091 in. thick) 6061-T4 anodized aluminum. The subchassis supports the power transformer, the two (2) power modules, and power supply filtering and electronics. The closed box-shaped structure is inherently rigid. Extra strength is added by the two (2) 'C' channel shaped covers that are manufactured from 11-gauge (0.091 in. thick) 6061-T4 anodized aluminum. The flat, machined front panel is manufactured from 4-gauge (0.204 in. thick) 6061-T4 anodized aluminum. Hardly a "basically flimsy box."

(Any 180/180-watt power amplifier we're able to lift without ruining our back is a basically flimsy box.—Ed.)

4. The power supply is a dual-voltage DC power source utilizing a power transformer manufactured to our design with 4% silicon transformer steel (type M-6). The better steel allows us to have a 30% weight reduction and over 100% improvement in load voltage regulation at a 100% increase in cost over the use of ordinary power transformer steel. The dual filter electrolytics are 24,000 microfarads each and are low-ESR, high-ripple-current, long-life computer grade (2,000 hr. specified life at 85° C) aluminum capacitors.

(Well, we happen to know that Dick Majestic is a good enough engineer to understand why the RAM 512 doesn't have a good enough power supply for a $1150 amplifier—not as good as the one in the Bryston 4B, for example—and we're willing to make that statement in 9-point Times Roman italic type with 1-point leading on a 13-pica measure, just in case irrelevant numerical specifications are the name of the game here.—Ed.)

5. The RAM 512 is kept cool by four (4) anodized extruded aluminum heat radiators. Each heat radiator has 244 square inches of surface area for a total radiation surface for both channels of 976 square inches. The four heat radiators, two (2) on each side of the amplifier, are placed in a vertically enclosed rectangular box. Radiant energy from the heat radiators is also absorbed by these vertical walls, thereby increasing the air flow and cooling. These top and bottom open boxes utilize the chimney effect to increase the air velocity across the surface of the heat radiators. This cooling system is quiet and efficient, and will allow the amplifier to be F.T.C. preconditioned at 1/3 power at 8 ohms without either of the two over-temperature protection thermostats shutting down the amplifier. This design allows the power amplifier to run cooler, therefore longer useful life will be attained from the amplifier without the use of a noisy fan.

(We consider this kind of cooling system, especially without a fan, to be inferior to large external heat sinks—and so do a lot of amplifier designers. But, you're right, the RAM 512 could still be a great amplifier if it were only for this.—Ed.)

6. As to your weight comparisons: the RAM 512 is mostly aluminum in construction. Aluminum is 34% lighter than its equivalent in steel. The use of better transformer steel allows further weight reduction, all at the sacrifice of increased cost; but then, filet mignon costs more per pound than ground round steak!

(Filet mignon construction but Gainesburger sound?—Ed.)

7. The RAM 512 will, with ease, produce 38 volts rms across 8 ohms at both channels and will also, with ease, produce 34.6 volts rms across 4 ohms at both channels, just as shown on the specification sheet. Typically, all 512(s) will also produce 38 volts across 4 ohms at 360 watts.

8. The square wave as reproduced by the RAM 512 is a replica of the input square wave, only deviating in slewing rate if the input square wave is slewing greater than 0.9 volts per microsecond. The RAM 512 exhibits a square wave with little ringing and no "peculiar kinks" at any power level from 10 mW to clipping, 20 Hz to 20 kHz and into reactive loads. Please note enclosed oscilloscope pictures taken on a new Tektronix 475 using a square wave source with 5 nanoseconds rise time. Note the smooth and linear rise and fall of the output signal slewing at 20 volts per microsecond, little to no ringing and without any "peculiar kinks".

(Most power amp designers would agree that a slew rate of 20 volts per microsecond is on the low side and may be partly responsible for the not-quite-first-rate sound of the RAM 512. On the other hand, we're willing to concede the possibility that only our particular sample put kinks in the square waves and that your pictures are of a more typical unit.—Ed.)

9. The phase shift at all frequencies between 20 Hz and 20 kHz should be no greater than 10° as stated on the spec sheet. Production 512(s) measure 7° at 20 Hz, 2° at 2 kHz, and 3° at 20 kHz. *(We measured more.—Ed.)* Input AC coupling creates the low frequency phase shift while the higher frequency phase shift is caused by the input RF attenuation network. Other minor amounts of phase shift result from the slew rate limiting and the output high-frequency phase compensation network.

I cannot change your opinion, arrived at after listening, but will say I'm sorry I didn't engineer in your preference in electronic coloration. *(What about that review in* StereOpus*? They, too, prefer electronic colorations? Everybody except you?—Ed.)* I can only suggest to you that you try the difference signal check. All that's necessary is to sum the inverted output signal with the input signal after readjusting for the amplitude difference and look at the resulting difference signal. *(Ah! But you have to look at both voltage and current differences!—Ed.)* It might prove educational to someone interested in audio electronics that most power amplifiers do not *just* increase the power level between their input and their output terminals. Loudspeaker loads cause some power amplifiers to do very undesirable and audible distortions when operated in the real world. Whereas the RAM 512, when operated in the same real world situation, will exhibit little or no distortions of the input signal.

Our audio industry is basically very small. We are quite friendly and co-operative people, and the propagation of hate does nothing to improve our sonic goals. *(Come on, Dick—you know we don't hate you. We just don't like the sound of your amplifier very much. Your problem may be that you can't* *distinguish between the two.—Ed.)* The consumer is just as dedicated to his purchase as we manufacturers are dedicated to the production of better audio equipment. We at RAM Audio Systems hope that what motivates you will soon be the same force that makes a manufacturer seek to improve this industry, improvement through a positive attitude.

Very truly yours,
Richard A. Majestic
President
RAM Audio Systems, Inc.
Danbury, CT

Where does all this leave us? Just here: The RAM 512, at $1150, doesn't sound as good as the Bryston 4B, the Futterman H-3aa, the Electrocompaniet, the $399 Audionics CC-2 (!) and any number of other amplifiers. We have yet to see a really favorable review of it in a noncommercial publication or meet a sophisticated audiophile who thinks highly of it. Thus the audible end result seems to bear out our own technical insights, not Dick Majestic's labored arguments.

—Ed.

The bitterest imbroglio so far, by a wide margin, concerns the Infinity QLS speaker system, as you can see from the following two letters.

The Audio Critic:

I did not answer the remarks concerning disparagement of the Watkins dual-drive woofer in earlier issues, as I do not believe anyone wins an argument with an editor in his own medium. However, after reading issue Number 4 of The Audio Critic, I feel the record should be set straight, so that owners of Infinity QLS speakers and those considering QLS will know the facts.

Concerning the uncomplimentary remarks of the American professor and the electroacoustician: My original disclosure article appeared in the December, 1974, issue of *Audio*. Of over two hundred letters received, only two were unfavorable—the two referred to by you. They were interesting, as neither contained any sort of technical complaint against the principle involved in the dual-drive woofer; the writers were close friends of each other, and both letters contained insulting remarks. I assumed the writers had a good laugh with each other, and I dismissed the letters as a prank. They apparently have no technical complaint, and why do you keep their names secret? In issue 4 you spend two pages condemning the writer of an anonymous letter against you, yet you create one against me by printing the smut from their letters against me without telling who wrote them.

(For moral obtuseness, this last remark takes the prize in our entire range of editorial correspondence to date. We know who said these things about your woofer article and you know who said them. We know what was said and you know it. We know it was disparaging and you know it. We know the academic credentials behind the disparagement and you know them. There's no disagreement and no concealed information between us! We're both *reluctant to name names because we* both *want to stay out of needless trouble. And you compare that to an anonymous poison-pen attempt to put **The Audio Critic** out of business by someone whose identity still remains unknown to all concerned! This from a man ostensibly dedicated to scientific logic and reason . . . —Ed.)*

Concerning the letter from Dr. Richard H. Small of the University of Sydney in Australia: Dr. Small was kind enough to do an analysis of the dual-drive woofer. He is one of the world's leading electroacousticians, and it was gratifying during the early stages of development of the woofer that he found it mathematically sound, and also that he was complimentary of it from an application point of view. The "knuckle raps" as you call them, concern how to explain the principle of operation—*not* the performance of the dual-drive woofer. The fact that you take things out of context does not speak well for your credibility.

(Maybe you *didn't fully understand Dr. Small's letter. What about that little quip, for example, about "taking more power from the amplifier without getting more acoustic output—in fact getting* less*?" That wasn't about the explanation of the principle but about the principle itself.—Ed.)*

Concerning the 2 dB advantage you mention in issue Number 4: The 2 dB figure is from Dr. Small's analysis with a *single* drive network, and this is a broadband efficiency gain (for the same shape of curve). As explained in the *Audio* article, the gain is 5 to 6 dB at low frequency with the double drive network—that is with an additional circuit to decouple the main voice coil at and below resonance. This is the circuit used in Infinity QLS speakers. *(But Dr. Small had some misgivings about the two-network driving system, didn't he?—Ed.)* I agree that an impedance peak and an amplitude response peak are two different things, however, in the case of the dual-drive woofer, response and impedance are *very* closely related. You are evading the issue anyway, and if you will measure an Infinity QLS, you will find that *both* frequency response and impedance are improved. *(Improved over what?—Ed.)*

You imply that the Watkins dual-drive woofer has no advantage when it comes to linear throw. A more linear throw *can* indeed be achieved. One simply makes the voice coil winding length longer, increases magnet to bring efficiency back up, and takes care of any overdamping at resonance with the low impedance voice coil to obtain the desired Q. Certainly, as you say, the woofer in the QLS needs a lot of power to drive it—it was designed that way. However, smaller QLS speakers use a slightly different woofer and have more modest power requirements.

In any event, the proof of the worth of a device is in what it can do, and the Infinity QLS speakers using the dual-drive woofer have been marketed for over a year now. During this time there have no problems with the dual-drive, and in fact I am very pleased with the acceptance these speakers have received. It offers a level of performance in a given size sealed enclosure that was previously unattainable. With reference to conventional design, it gives more extended low frequency response, higher efficiency, or a smaller enclosure can be used. In all cases the impedance peak and the phase shift at resonance are drastically reduced, and this presents a less reactive and more resistive load to the amplifier.

You say you sell information and advice. For the price you charge you should at least offer correct information.

Sincerely,
William H. Watkins
Kingsport, TN

***The Audio Critic** never called the Watkins woofer worthless. It merely raised an eyebrow about the claims of a free lunch obtainable from such a design. And it reported that a distinguished professor of electroacoustics had dismissed the woofer with contempt.*

What we do believe after careful consideration of the subject is that the Watkins woofer represents a rather trivially conceived design trade-off, namely higher efficiency in its upper range of frequencies than would be possible without the dual drive—all other things remaining equal—but at the cost of chewing up a great deal more amplifier power at the lower frequencies around the system resonant point. That's all the invention essentially is; there's nothing more to it, and that's what all the shouting is about. It's possible to duplicate the frequency response, time response, efficiency, distortion characteristics, etc., of any Watkins woofer with a conventional single-voice-coil woofer simply by accepting a somewhat different set of trade-offs. Big

deal. How about a real invention, like a large-area electret woofer, for example?
—Ed.

The Audio Critic:

We were naturally surprised at some of the comments about the Quantum Line Source loudspeaker made by The Audio Critic in its last issue. However, it would serve no purpose to debate Mr. Peter Aczel's subjective reactions, differing as they do from those of other reviewers.

(The subjective reactions published in ***The Audio Critic*** *are those of at least three persons, more often of five or six, never just the Editor's. As for other reviewers,* Sound Advice, *for example, called the QLS "dreadful" and suggested that the whole idea be scrapped and started all over. We were much kinder. —Ed.)*

We feel bound to write this letter because of the distressingly pejorative way in which Mr. Aczel refers to people of considerable ability, whose efforts have made real contributions toward advancing the state of the art of high fidelity. *(Name one.—Ed.)*

For someone so willing to be pejorative, Mr. Aczel should at least be more consistently correct in the factual information he presents, and more careful in his use of technical terms. The latest issue of The Audio Critic shows carelessness and inaccuracy which does not bode well for the publication's future credibility.

For example, Mr. Aczel criticizes Infinity for naming one of its speakers the Quantum Line Source, on the grounds that there are, in fact, two parallel line sources. Mr. Aczel does not go on to explain exactly what the disadvantage of this configuration is—it is, of course, primarily that there will be some interference effects. The interesting thing is that elsewhere in the same issue (page 24) he himself calls the Rogers LS3/5A (an excellent speaker, we agree) a "point source". It is, of course, not a point source—because there are two sources of sound which will interfere with equally deleterious effects. Elsewhere in the same issue, the Beveridge (also an excellent speaker) is called "a *true* line source"—when it cannot operate as a *true* line source by virtue of its dimensions. Of course, neither can the QLS. But for someone who is so frequently critical of other people's uses of language, Mr. Aczel is uncommonly careless with his own.

(We never said that the Infinity QLS is made up of two parallel line sources. You're the one who is saying it, and of course you'd have a better speaker if it were true. The fact is, however, that the QLS is comprised of individual drivers radiating spherically in the nearfield. This generates a constructive/destructive interference pattern spread out over the sizable dimensions of the system. The end result is that in the farfield, where planar radiation exists, the sum of the energies generated by the system does not resemble that of a true—or quasi—line source. A line source radiates a uniform energy field—i.e., controlled interference—in the nearfield. Thus a line source already starts in the nearfield with controlled line radiation and not with a bunch of hemispherical radiators whose vector sum is simply not a line. The Beveridge is far ahead of the QLS in this respect. As for the Rogers, it's obvious that you're just trying to hassle us. You know as well as we do that the few inches between the two drivers of the LS3/5A shrink to virtually nothing—i.e., a point —from the acoustic perspective of a listener ten or more feet away. Of course, to a very *short listener a* very *short distance from the speaker it might not appear as a point source.—Ed.)*

In the review of the Rogers LS3/5A, for another example, the reviewer implies that this most excellent loudspeaker was designed by paying "special attention to the time and phase characteristics" which, in fact, it was not. Mr. H.D. Harwood, who was heading the BBC team that designed this loudspeaker, has stated several times that he totally ignores these parameters in the design of loudspeakers because he regards their effects, at best, secondary compared to other problems. The reviewer then goes on to imply that the fundamental resonance of the LS3/5A is at 130 Hz with a Q of 2. If he had measured the impedance curve, he would have found that the fundamental resonance is, in fact, around 80 Hz, corresponding closely to the -3 dB point and a Q of 0.7. This is probably why his visitors told him that the bass sounded just great—although limited in extension of course. (An indication of the disregard for time alignment in the design of this system is the position of the woofer, which is actually mounted on the back of the front baffle).

(Intentionally or not, the Rogers does reproduce pulses quite accurately, albeit somewhat off axis. That's why it sounds best from that same angle off axis. Many advancements in technology have been serendipitous. The point is, would the Rogers sound as good as it does if it couldn't reproduce a pulse at all? When it comes to the Q of 0.7 you're in deep mathematical trouble, fella. Q = 0.707 means a maximally flat—i.e., Butterworth—contour, with the amplitude response of the system 3 dB down at the resonant frequency with respect to the nominal bandpass. Maximally flat means just that: no ripples whatsoever in the response. Now, if you measure a second-order system using carefully calibrated equipment in the manner described by D. B. Keele, Jr., and find that the amplitude response shows a peak of 6 dB near the system resonance, then drops to a steady level 6 dB below that peak as the higher frequencies are continuously swept—well, that just isn't a Q of 0.707, Arnie. It's a Q of 2 and that's that—period, end of story, not negotiable. Ask anyone who has studied the subject. It's perfectly true, of course, that the Q of a system can be derived by analyzing its impedance characteristics. In fact, with a low-Q system, it's the only way. But in the case of a relatively high-Q system like the Rogers, nearfield amplitude measurement will quite accurately indicate the value of Q.—Ed.)

The faith that Mr. Aczel is now placing in his pulse tests is extraordinarily naive. The information that one can obtain from such tests is very limited and, doubtless because of other much greater imperfections in loudspeakers, there is little relation to perceived quality. The Rogers speaker is one example of this—we ourselves have found a number of other examples. The QLS is the case in point. The first few of these speakers that we made, and the original prototype, had the woofer mounted forward of the remainder of the drivers in order to produce a more accurate pulse response. We discovered after a while, however, that whatever the pulse response benefits of this arrangement, they were masked by other coloration caused by practical mounting arrangements. We have encountered this same problem several times since—i.e., the effects of staggering drivers for time alignment can introduce worse colorations than they were designed to reduce.

(Well, let's see who some of the other "naive" practitioners are who put their faith in pulse tests and related measurements in the time domain. Richard C. Heyser, for openers. Berman and Fincham of KEF, the 3-D response display people. Harold Beveridge, who designed his speaker almost exclusively with the aid of time-domain tests. Are we in bad company? As Andy Rappaport points out in his letter elsewhere in this column, audio is nothing more than amplitude variations with respect to time. We all live, and hear, in the time domain. Of course, if you use time-smeared program material—as your remarks on cartridge alignment below seem to indicate—then you won't hear the difference between speakers with good and bad time response.—Ed.)

We believe in scientific methods in the design of speaker systems just as much as Mr. Aczel—but at the present time there are a number of reasons why any individual measurement technique offers only a very limited guide to the overall perceived quality of the system. The main one, of course, is that because all loudspeakers are so far from being truly accurate transducers, a considerable subjective element of design must occur in terms of making a judgment as to which particular set of imperfections is least objectionable.

Coming onto Mr. Aczel's comments on tone arms, we are surprised at his contention about the necessity for very accurate setting up of lateral tracking error. There are two separate issues, it seems to us—first, the audibility of small amounts of lateral tracking error. We have yet to hear this successfully demonstrated under controlled conditions but are willing to accept it may be audible. It is the second issue which interests us—if a cartridge is slightly misaligned, all that happens is that the position or positions on the disc at which there is zero error, change —i.e., there is now zero error on different parts of the disc than previously. (Unless, of course, with unparalleled ingenuity the arm is arranged to move as the record is playing.)

(Wow! You really blew it there. Alignment for optimum tracking geometry means, by definition, minimizing the value of α/R throughout the record —α being the lateral tracking error at any given point and R the radius at which it occurs. That the solution of this mathematical problem results in two nulls, where there's no tracking error at all, is purely incidental; the goal is to keep the peak values of α/R as low as possible. If α/R could be further minimized by not going through the nulls, that's the way it would have to be done. But the mathematical nature of the beast is such that there exist those two nulls and—here's the point—their position is fixed, as long as the maximum and minimum radii are specified. Move the nulls and you've increased the value of α/R somewhere between the maximum and minimum radii. We find it tragic that this should have to be explained to an engineer who sells tone arms for a living.—Ed.)

Then again (page 52) the writer shows his lack of knowledge in his comments about the importance of antiskating compensation. The Audio Critic should make itself aware of some work done in England on the wide variations encountered in the frictional force between the record and stylus (which, of course, causes the so-called skating force). It has been clearly demonstrated that the frictional force varies with a number of parameters including the stylus itself, the record material, the temperature, and the modulation of the record groove itself. These variations are not just small—they are the order of several times. It is therefore absolutely impossible to set bias force accurately for anything other than one particular and very limited set of conditions—because the force actually varies while any given record is being played.

(You're wrong. With a given stylus, a given tone arm and a given vertical tracking force, the antiskating compensation can be set permanently for all practical purposes. We shall go into the theory behind this in a future issue; this parenthesis is hardly the place for a treatise. In any event, the simple test we describe on page 55 of our Number 4 issue will tell you whether or not the compensation is correctly set.—Ed.)

Then again the magazine does not seem to be aware of the benefits of low mass, judging from its comparison of the Grace G-707 with the Infinity Black Widow. Of course, the Denon cartridge that you were using is not a very compliant one. More modern cartridges need the very lowest-mass arm if they are to work correctly—some of these cartridges are even too heavy for their own compliance and theoretically, therefore, would need an arm of negative mass. This aspect seems to have been completely ignored. We also need to comment about the review of the Black Widow arm per se—the bearings are not "prone to jitter" under actual conditions of use. Indeed, the geometry and the forces involved prohibit it.

(It just so happens that the best of today's cartridges are not the ultracompliant ones and therefore have no need for an ultralow-mass arm. See the cartridge reviews in this issue. As for the bearings of the 'Black Widow' arm, if they aren't tight this *way, they aren't tight* that *way. No amount of sophistry will make them as positively located and as jitter-free as those of the Grace G-707. But then a little shift here and there in the stylus-to-groove relationship is of no importance, is it Arnie?—Ed.)*

More facts. Mr. Aczel mentions in the QLS review that the midrange dome used in the QLS is a Peerless driver. In fact, it is not, and originates in Germany from another member of the group, MB Microphonbau. (*So it's a Dodge instead of a Plymouth. Sorry about that, but it's hard to tell those Chrysler products apart.—Ed.*) The holes in the dome do not act like a Helmholtz resonator—they are much too small to do that effectively. The holes improve the transient performance of this driver, at the expense of low-frequency extension. Mr. Aczel then goes on to say that Peerless also supplies the 5" midbass driver—it does not, as a matter of fact. These issues in themselves are not, of course, particularly important, but they do indicate the haste and eagerness of Mr. Aczel to speak out without checking his facts first.

(If it isn't the Helmholtz principle that makes the holes increase the output of the dome, pray tell what it is? The California sun shining through them? Obviously, energy is being derived from the presence of these holes punched in the diaphragm, suggesting rather strongly a harnessed resonance used to reinforce the output from the front surface of the dome. The dome is vented, *so to speak, to increase output over a narrow band of frequencies, and this method most certainly relies on Helmholtz resonator behavior. The claimed improvement in transient response flies in the face of science, we're afraid. Given a system defined by a transfer function, increasing the complexity of the system—all other things being equal —will degrade rather than improve transient behavior. In this case, the midrange dome is essentially a second-order device whose behavior can be mathematically specified by a second-order transfer function. This transfer function specifies both the magnitude and phase of the system. Increasing the complexity of the system by punching holes in the diaphragm raises the order of the system from second to third or fourth. Even if the characteristics of the system are preserved—say, by maintaining maximally flat behavior—the phase response is substantially degraded, suggesting in the time domain a poorer, rather than improved, transient response.* ***The Audio Critic****'s tests, however, indicate a severely peaked output where the holes become effective, and that means a substantially worsened transient performance. As for the 5" midbass driver, whether Plymouth or Dodge, it rings like an alarm clock and should be pulled out of the system and stomped on.—Ed.)*

As to the reviewer's subjective conclusions on the sound quality of the QLS, we cannot usefully comment, except to say that insufficient attention may have been given to setting up the system, and to the associated components. It certainly is surprising that his findings are so widely at variance with those of other independent reviewers, and this would seem to indicate that something else was wrong. His comments about the stereo image, for example, could result from the use of a tone arm of too high mass.

(Okay. Name one audio professional—just one—who has visited ***The Audio Critic****'s laboratory and listening*

room lately, and then felt that better procedures and cleaner program sources were available elsewhere.—Ed.)

We find Mr. Aczel's gyrations on the Watkins woofer to be quite amazing. Having first dismissed it completely in an earlier issue, in his review of the QLS Mr. Aczel attempts to backtrack. Having originally condemned the Watkins design out of hand altogether, he now admits, as Dr. Small has said, that the concept does indeed allow an improvement in low frequency response. But he is determined it appears not to lose face and so has attempted to denigrate the design. An improvement of 2 dB in output, other things being equal, is indeed far from negligible. It means that a closed box system, with its advantages over vented systems of high mechanical impedance at subaudio frequencies, and a gentler cutoff slope, can nevertheless be directly competitive with such vented systems in terms of efficiency. Moreover, the extra output is obtained and the damping is increased at the same time. We should perhaps quote directly from Dr. Small's letter to William Watkins, to which Mr. Aczel refers, " . . . my congratulations to you on devising a significant improvement for loudspeaker design." Hardly "condescending"!

(*Yup. Dr. Small is a nice guy and didn't want William Watkins to feel that the weaknesses pointed out in his design made it totally worthless—which, incidentally, we never said, either. We refuse, however, to pursue the subject of the Watkins woofer any further than our reply to Watkins's own letter above. That's how we see the whole thing and that's all, folks. So, from here on down, we'll just let Arnie noodle with his favorite theme to his heart's content and with impunity.—Ed.*)

We agree the Watkins design does not have any advantage in terms of power handling ability over an ordinary sealed box, other things being equal. We never claimed it did. Of course, reflex and 'transmission line' enclosures make use of an acoustic resonance to reinforce the bass—thereby increasing the power ability, other things being equal—but these designs have another very much more serious disadvantage, which is that the mechanical impedance below the audio band is very low, allowing the cone to make large displacements. In practical circumstances, this causes far more distortion than a greater cone movement within the audio band. We are surprised that Mr. Aczel managed to overload the QLS woofer—because this is not a problem that we have come across. It is possible that he clipped his amplifier, because the acoustic effects are sometimes difficult to distinguish.

We agree with Mr. Aczel (and now come to that Dr. Small letter) that the Watkins principle is difficult to explain—what Mr. Aczel called "professorial knuckleraps" in Dr. Small's letter are, in fact, connected with Mr. Watkins' explanation of the principle—not the principle itself. Indeed, Dr. Small himself says, "unfortunately I cannot offer any simple cure-all suggestions for presenting a better explanation." If Mr. Aczel has any suggestions as to how we may explain the principle more simply, we would be delighted to hear of them. However, he should recognize that there is a difference between the validity of a design principle, and the ease with which it can be explained in simple technical terms. He is putting up a smoke screen to conceal his own lack of credibility by debating the method of explanation—the fact is that William Watkins indeed has a useful and valid contribution to make and this is at direct variance with Mr. Aczel's original comment, which, as usual, was as hastily prepared and inadequately researched as it was pejoratively stated.

Yours sincerely,
Arnold Nudell
President
Infinity Systems, Inc.
Canoga Park, CA

Our subscribers may be wondering why we reproduced this letter in its petulantly long-winded entirety and why we made an attempt to refute it point by point. Well, for one thing, we want the audio community to understand once and for all that a critiqued manufacturer does have the opportunity to argue with us in this column, even though we don't solicit such an argument before publishing the review. At the same time, the letter has some consumerist value, since Infinity is a rather heavily promoted audiophile brand and the consumer has a right to know just what kind of technical thinking and preparedness is behind those products. Since the chief technologist of the company is President Arnie Nudell himself, you can form your own conclusions from the above. —Ed.

In Your Ear

"Hey, they top-rated my power amp!"

"These people know what they're talking about."

"Ah, they really like my tone arm!"

"You know something? This is by far the best audio review. The others aren't even close."

"What? They think my speaker is terrible!"

"Ignorant nincompoops! Who gave them the right to review audio equipment, anyway?"

Calderhead

The Credibility Crisis in Equipment Reviewing

By Peter Aczel
Editor and Publisher

All sorts of publications are reviewing audio equipment, but on whose review would you confidently stake a four-figure buying decision? We discuss some criteria, perspectives, caveats, and consequences.

Our sequential numbering of topics is continued here merely for easy reference, without any suggestion of a strict serial relationship to previous articles.

* * *

20 When we started **The Audio Critic** early in 1977, we had something fairly simple in mind. We felt there was room for an audio publication that would combine the laboratory disciplines and editorial professionalism of the hi-fi slicks with the golden-ear hypercriticism and commercial independence of the "undergrounds." After five issues we have a totally different perspective.

Today we see the entire process of consumer communications in audio, from both commercial and noncommercial sources, in the midst of a credibility crisis, and we see ourselves not so much as the providers of a worthwhile "second opinion" but rather as the only confident and habitual defenders of reality—i.e., the laws of nature and the evidence of comparative testing—outside the professional journals. (Always excepting Richard C. Heyser, who on rare occasions visits the commercial hi-fi press from his faraway observatory of ultimate realities.)

* * *

21 The credibility crisis exists in two more or less independent areas, which nevertheless overlap in the most fundamental sense: (1) technical background information and (2) reference values in critical listening. Both of these qualify as national disaster areas.

The first is the result of the utter comfort that both makers and reviewers of audio equipment seem to take in the most superficial poptech assumptions and generalizations, thereby preventing the ordinary consumer from ever coming face to face with the basic, first-year-physics facts of life.

For example: What commercial or noncommercial audio publication has ever made reference to the fact that there exists an attainable mathematical optimum in woofer design and that nearly all woofer designers are ignoring it? Or that there exists only one mathematically correct number defining the maximum dB's of feedback allowable in a given amplifier circuit and that nearly all amplifiers exceed it? Or that the most common RIAA equalization network used in preamplifiers can't possibly be 100% accurate, even with tight-tolerance resistors and capacitors? We don't see how anyone can review audio equipment, even for a nontechnical audience, without the support of this kind of specific background information, but of course there's no sign of it anywhere in print.

As for reference criteria in listening tests, how many regularly published reviewers religiously align their cartridges and tone arms for optimum lateral *and* vertical tracking geometry? None that we know of. And if they don't, how do they presume to arrive at a valid opinion about the sonic character of any piece of equipment through which their degraded reference signal passes? And how many reviewers use a reference speaker that can reproduce pulses of various widths with a semblance of accuracy? We suspect that the vast majority of audio practitioners, whether on the manufacturing or the reviewing end, have never really experienced the startling clarity of a system with optimized time-domain characteristics from stylus tip to speaker diaphragm. Seasoned audio people who visit our sound room are invariably astonished, although we don't do anything there that they themselves couldn't duplicate if they ordered their priorities as we do. (Which is, of course, where the information crisis meets the listening crisis.)

* * *

22 An inevitable consequence of our departures from conventional testing and reviewing is that we need time—more time than we had anticipated before we published our first issue. (Though not as much time as others appear to need to cover far less ground—but we've been through that before.)

Think about it. When the commercial magazines receive a piece of equipment for testing, they weigh it, count its knobs, measure its frequency response, and you know the rest. When the "undergrounds" get one, they listen to it—with their tone arms aligned according to the incorrect instructions that came in the box and everything else similarly shipshape—and find that the upper midrange is whitish and the lower highs insufficiently liquid. Compare either approach with what you've seen in our speaker survey, for example, and you'll begin to understand why our originally projected bimonthly schedule didn't quite work out. It takes two experienced people a full working day—sometimes even two days—just to analyze *one* speaker in the lab, and that doesn't include the endless listening evaluations, the write-ups and the editorial work.

Mind you, we're not complaining. We like it this way. We'd rather publish six credible issues in seventeen months than six questionable ones in twelve. But every once in a while we get a letter suggesting that somehow we're perversely withholding information that we should have spewed out in print months ago. We'd hate to think that these letter writers would actually be happier with whitish-and-liquid every two months, like clockwork . . .

* * *

23 Speaking of credibility, the oldest of the "underground" audiophile reviews has gone 100% commercial. In addition to hi-fi store advertising, which we never thought was entirely noncorruptive to begin with, it now also runs manufacturers' advertising. And, guess what, all the manufacturers who have an ad in the latest issue are doing very well, thank you, in the "Recommended" listings!

In the same issue, the editor takes a rather aggressive swipe at **The Audio Critic**—not by name, but by specific reference to the views expressed under topic 13 in our editorial series, so it might just as well have been by name. He observes that we foolishly (foot-in-mouth is the friendly image he uses) discriminate between "accuracy" and "musicality." It's easy enough to go back to topic 13 and prove that we were saying just the opposite—that inaccuracy may try to masquerade as musicality but in the end will always be musically bested by accuracy—but the editor's reading disability isn't our concern here. What we find remarkable is that a publication with a triple credibility gap—namely, manufacturers' advertising, lack of realistic technical rationales, and time-smeared reference material via "Recommended" moving-magnet cartridges—should take it upon itself to challenge the credibility of **The Audio Critic**.

They say that the right enemies are just as much a sign of success as the right friends. If that's really true, we've arrived.

Sophisticated Speaker Systems, Large and Small: Continuing Our Survey

By the Staff of
The Audio Critic

Part II: In which a new small box goes to the head of its class, a $3000 audiophile legend falls flat on its face, a "best buy" turns out to be even better, and our reference speaker remains unchallenged.

We have very little to add at this point to the general discussion that preceded our first batch of reviews in this survey (see our Number 4 issue, pages 18 to 21). Our assumptions and our methods remained the same when we tested the speakers reviewed below.

The only further insights we have gained since the first go-around have to do with the specific sonic quality of certain measurable speaker characteristics. We have zeroed in a bit more precisely on a number of frequently observed anomalies.

For example, a steely, ear-piercing quality has generally very little to do with frequency response. It can nearly always be traced to ringing in the 2 kHz to 5 kHz range, where the ear is most sensitive. A tipped-up amplitude response at the higher frequencies may sound overbright but doesn't "burn" the ear unless there's considerable ringing, identifiable with tone bursts.

Poor time response, as indicated by the inability to reproduce a recognizable pulse, is usually heard as a homogenized blending of inner textures and a lack of delineation of spatial detail. It is *not* irritating or unmusical if the amplitude response is reasonably smooth and ringing is moderate or absent. For that reason a lot of people throw their hats in the air and shout "State of the Art!" when they hear silk-smooth, wide-range audio without noticeable graininess or grit, unaware that the requirements of accuracy go well beyond that. Genuinely lifelike sound reproduction is characterized by an open, focused, see-through quality that cannot be achieved without proper attention to time response.

Underdamped woofers also have a characteristic sound, a whompy heaviness that passes for "powerful" bass in some circles. We continue to find most woofers delinquent when it comes to properly calculated damping; their Q is unnecessarily high. The whomp can sometimes be made more tolerable to the ear by placing the speaker away from all boundaries, i.e., far from the walls and off the floor.

(Admittedly not a practical solution under all circumstances.)

The honky midrange quality that degrades the performance of so many expensive speakers is also due mainly to ringing, anywhere from 250 to 800 Hz. Often the woofer is required to reproduce these and even considerably higher frequencies, without any discernible attempt on the part of the designer to control cone behavior above the "piston" range.

In general, it's safe to say that pressure amplitude response (what is loosely referred to by audio people as "frequency response") is no longer the limiting factor of speaker performance as it used to be years ago. Today's main shortcomings are undamped oscillations (i.e., ringing), severe trade-offs between accuracy and dynamic range, and poor time response.

The Audio Critic as a consumerist force.

It may be interesting to our subscribers to learn that Part I of this speaker survey sent at least five speaker designers back to the drawing board. We have reliable reports on this in each instance. And we find the situation both encouraging and discouraging.

Encouraging because it shows that there are designers who are more interested in how their product will perform tomorrow than in how infallible their views were yesterday. Of such stuff is progress made. It also proves that consumerism can work even in the frenzied, ego-tripping world of high-end audio.

On the other hand, we consider it discouraging that our tests were news to so many people who earn their daily bread with audio engineering. **The Audio Critic** has so far developed no original concepts in testing. We may look into a lot of things that others neglect, but everything we do is based on information and procedures obtainable in any decent engineering library, most of it in the last 14 or 15 volumes of the *Journal of the Audio Engineering Society*. We don't own a single laboratory instrument—or audio component—that can't be purchased or at least borrowed by any speaker company. The ears of our staff are well-schooled but in no way phenomenal. And the laws of nature are the same at our address as elsewhere. So how come our test results are a surprise to so many professionals?

Let's proceed to our reviews before we get too worked up over the industry's mentality.

Beveridge 'System 2SW'
(follow-up)

Harold Beveridge Inc., 505 East Montecito Street, Santa Barbara, CA 93103 (note new address). Beveridge Cylindrical Sound System, Model 2SW, $6000 the pair (new price, including HD subwoofers, plug-in power amplifiers and CM-1 control module). Unlimited warranty on all parts except tubes (one year); five-year warranty on all labor, including sonic updates. Tested #233 and #234, on loan from manufacturer.

Now that we have lived with this system for a number of months and used it as our principal reference speaker for evaluating preamps, phono cartridges, etc., we believe more than ever in its overall superiority to other designs and at the same time are frustrated more than ever by its obvious deficiencies.

From about 100 Hz on up, the Beveridge is simply more accurate than any other speaker known to us; it reveals more about the signal fed into it than any other and just sounds more like real life. No question about it. We now feel that Harold Beveridge's idea of creating a coherent cylindrical wave front out of a planar wave by means of a limited number of wave guides (his amazingly simple "acoustic lens") is one of the major conceptual breakthroughs in the recent history of audio; furthermore, on the purely practical plane, his electrostatic transducer "sandwich" is the best in the business. These two points of superiority give the Beveridge system a substantial advantage right up front. But there are a few flies in the ointment.

As we pointed out before, the woofer is far from SOTA; a $6000 system (that's the latest price) ought to have tighter, deeper, more awesome bass. A 12-inch driver in a sealed box just can't do the ultimate job, even with the Q and all other parameters optimized (which they don't appear to be in this case). What we'd like to see here is something like the sheer air-moving capability of the 24-inch Hartley, combined with the damping characteristics of, say, the Cizek or the Fundamental Research. After all, the woofers alone cost $500 apiece if you order them separately.

As for our complaints about the SPL limitations of the Beveridge, we've just made an interesting discovery. The sheer noise-producing capability of the system is much greater than we would have believed on the basis of our original tests with musical reference material. For example, the railroad sounds on Side Two of *The Power and the Majesty* (Original Master Recordings, Mobile Fidelity Sound Lab MFSL 004) can be played loud enough to drive everybody out of the room without any obvious distress signals from the speaker—but then we must admit that our ears aren't attuned to subtle distortions in locomotives. On the other hand, the New Haven Brass Quintet on Side D of Volume Four of the Mark Levinson Acoustic Recording Series begins to sound edgy and clipped when played at a you-are-there level, whereas a couple of dB below that it's absolutely gorgeous. This leads us to believe that the frustrating dynamic limitations of the Beveridge may conceivably be due to nonlinearities in the amplifier rather than to the inherent SPL ceiling of the electrostatic panels. If we're right—and of course we may just as easily be wrong—this is very good news, since the amplifier could be much more easily redesigned than the electrostatic panels. Harold Beveridge is well aware of these considerations and will undoubtedly come up with an answer—if there is one.

As far as the 8 dB peak at 76 Hz is concerned, it was a prototype boo-boo that has been fixed in the production model. Our early samples were modified accordingly and we can now give them a clean bill of health. The peak is measurably and audibly gone.

Which still leaves us with a speaker that we can neither exalt without a number of reservations nor replace with a better one. It isn't as good as it ought to be; it's merely the best there is.

Canton HC 100

Adcom, 114 East 32nd Street, New York, NY 10016. Canton HC 100 miniature speaker system, $180 the pair. Five-year warranty; manufacturer pays return freight. Tested #002673 and #002674, on loan from manufacturer.

Teeny-weeny speakers with an audiophile appeal seem to be a German specialty; this one is Canton's answer to the Braun 'Output C' minisystem. The HC 100 is only very slightly larger, its longest outside dimension being 18.5 cm (7¼ in), but it has an irregular shape with a sloping front grille and is intended to be deployed horizontally. The driver complement consists of a 4" cone and a 1" dome, crossed over at 1700 Hz.

We happen to be somewhat obsessed with this type of speaker, always hoping against hope that one of them will turn out to be absolutely superb above 200 or 250 Hz and could then be used with a hidden subwoofer as an almost invisible system of near-SOTA performance. No such luck in this instance, even though the HC 100 has some very commendable qualities, just like the larger Canton LE 400 reviewed in Part I.

The system resonance of the sealed-box woofer (if you can call a 4-incher by that name) is in the neighborhood of 200 Hz, and the system Q is approximately 2, which of course is too high—but all minisystems try to fake extended bass response that way. Between 500 Hz and 20 kHz, however, the response of the system is spectacularly flat, very similar to that of the LE 400. The Canton people seem to have this part of the problem pretty well figured out. What's more, our tone burst tests proved the HC 100 to be remarkably free from ringing.

Our pulse tests, on the other hand, indicated that all was not well in the time domain. The HC 100 appears to be totally incapable of reproducing pulses of any duration with even a semblance of coherence. And that makes it a lot less interesting to the audio perfectionist than it might have been.

The sound? Exactly what you'd expect. Very smooth, sweet, balanced and nonfatiguing, remarkably wide in range, but without any real focus, air or spatial detail. You might as well be listening to mono. The Braun Output C is somewhat better in this respect, even though it rings more and is less flat than the HC 100. One day, either Braun or Canton will get this whole act together, but it just isn't happening yet.

We hear that one of the preferred uses of the Canton HC 100 is as a Mercedes-Benz or BMW custom speaker. Since the drivers of these cars are already satisfied in the time domain, so to speak, that probably works out very nicely.

Dayton Wright XG-8 Mk 3

Dayton Wright Associates Limited, 350 Weber Street North, Waterloo, Ont., Canada N2J 4E3. XG-8 Mk 3 Series 3 full-range electrostatic loudspeaker, $2995 the pair. Five-year warranty. Tested #2422A and #2422B, with Model ST-300A matching transformer and bias supply #0723, on loan from manufacturer.

Whew! What a letdown. Here's the speaker that epitomizes exotic audiophile equipment: completely offbeat design, limited production, stratospheric price, fanatical cult following, ritualistic explanations. And the sound isn't even pleasing, let alone accurate.

We must admit that every experienced audiophile who heard our pair of Dayton Wrights remembered the speaker as having sounded better in earlier versions. We had the very latest Mark 3 Series 3 model, made entirely under the new Leigh Systems regime, after Mike Wright's departure. There was certainly nothing defective about our samples; someone at the company had warned us about the possibility of overinflation of the "gas bag", but that was definitely not the case. (The 10 full-range electrostatic cells of the XG-8 are sealed in a special gas environment that permits much higher operating voltages than air.) We're fully satisfied that what we tested was representative of current production; what the speaker *used* to be is a moot point.

This particular pair of Dayton Wrights sounded honky and unclear in the midrange, aggressively hard, and almost instantly fatiguing. The only good things we heard were quite decent bass for a wooferless dipole speaker of one square meter (11 sq ft) area, plus excellent dynamic range. (Piano recordings could for once be played at the same level as a live piano.) But the colorations of the speaker were too severe for us to take it seriously as a reference-quality system.

Before we go into the specifics we want to reassure the cultists. We had the Dayton Wrights plugged in and charging for more than a week before we even went near them. We then listened to them, measured them, and listened to them again. At length. After another month of uninterrupted charging, we tested them again. To those who believe that even this wasn't enough, that it takes six months of continuous charging before a Dayton Wright is ripe for evaluation, all we can say is that we were looking for a good loudspeaker, not a good cheese.

Our measurements revealed some serious problems in both the frequency domain and the time domain. The bass response of a dipole is hard to measure (see our comments under the Koss Model One/A in Part I); we estimate that the -3 dB point of the Dayton Wright is at 45 Hz, with some roughness in the 40 to 60 Hz region. From there on up we observed a constantly rising response to a peak at 700 Hz, with a falling response thereafter. In the 14 to 15 kHz region there's another bad peak, which comes from the "piezoceramic ultratweeter" used at the highest frequencies. (We don't believe that such a device belongs in a full-range elecrostatic system, disagree with the rationale given for it in the Dayton Wright literature, and would censure it in greater detail here if it were the speaker's only flaw.) These two peaks dominate the sonic "signature" of the XG-8 Mk 3. The 700 Hz peak, which persists regardless of the angle of measurement, has an amplitude of approximately 7 dB as compared to the surrounding frequencies and is almost certainly the main cause of the speaker's honky coloration and unclarity in the midrange. The tweeter peak has an amplitude of 10 dB with a fairly low Q (i.e., its base spreads over a fairly broad section of the spectrum) and is the probable cause of the zingy quality of the highs.

Even more significantly, severe ringing occurs at both of these peaks (namely at 700 Hz and 14.7 kHz), as well as at 1.8 kHz and 4.5 kHz, where we observed no comparable amplitude anomalies. In general, the speaker is highly prone to ringing throughout the audio range, with many other trouble spots where tone burst reproduction is rather poor, though not as bad as at the four frequencies mentioned. Occasionally the ringing drops to reasonable levels, but on the whole there can be no doubt that the system's integrity is ruinously compromised.

Exploring the XG-8 Mk 3 with widely separated pulses revealed that the rise time in some instances is quite a bit slower than one would expect of an electrostatic transducer, and time smear was also evident at a number of frequencies. Not a glorious picture, all in all.

The way we see it now, the main and perhaps only permanent contribution of the Dayton Wright elecrostatic system to the advancement of audio is the technology of operating electrostatic panels in a sealed, gas-filled environment. By operating at voltages far higher than the ionization threshold of air, the Dayton Wright permits loudness peaks and a dynamic range comparable to, say, the Pyramid Metronome's. That's about it. We can't think of anything else to admire in the speaker, especially in view of its $3000 price.

Oh yes. We did use the new stands that raise the Dayton Wrights off the floor and tilt them slightly backwards. We didn't, however, drive them with the Threshold amplifier as the cult demands, but with a measly Bryston 4B. Perverse, aren't we?

(Those who feel that the Bryston causes peaks and ringing in the Dayton Wright, whereas the Threshold does not, are invited to send us their revisions of the laws of physics for immediate publication.)

DCM 'Time Window'

(Improved)

DCM Corporation, 2275 South State Road, Ann Arbor, MI 48104 (note new address). 'Time Window' floor-standing loudspeaker, $660 the pair. Five-year warranty. Tested #1853 and #1864, on loan from manufacturer.

A number of worthwhile improvements have been made in this already remarkable speaker, so that a reevaluation is in order.

The main visible change is the switch to Mylar domes in the tweeters (with mixed results, as we shall see), but the overall sonic character of the speaker is sufficiently different to indicate that other engineering details have been retouched as well. You can assume that all units with a serial number above 1600 incorporate these refinements.

The improved version of the Time Window sounds even more open and transparent than the original, with smoother and better-defined highs, and hardly any whomping or thickness in the upper bass and lower midrange. It's simply a more natural sound, immediately apparent in an A-B comparison.

The amplitude response of the speaker is now extremely smooth between 200 Hz and 14 kHz, within ±3 dB we'd say. At 14 kHz there's a rather nasty tweeter peak that definitely wasn't there before; its amplitude varies with the measurements angle (9 dB on the axis of the dome, 6 to 8 dB in the composite response curves), but it's always present and cannot be "homogenized" out by moving the microphone. The tweeter response drops precipitously after the 14 kHz peak; at 20 kHz it's good-bye and gone (down to -14 dB). Interestingly enough, this anomaly has relatively little effect on the speaker's audible performance, probably because the Q of the peak is quite high (i.e., the spurious energy is contained within a rather narrow band of frequencies). Tone bursts confirm the reality of the peak, exciting severe ringing at 14 kHz, even though the rest of the audio range is quite as free from ringing as before.

On the low end, the tuning of the vented enclosure is still highly suspect, although it appears to have been changed slightly. The vent still doesn't fill in correctly the null in the driver response (in this case at 32 Hz), and the composite bass response is still quite lumpy—we'd call it ±5 dB from 40 Hz on up. On the other hand, the woofers are well-behaved under transient excitation; they shut up when the signal stops. In this respect we see (and hear) some improvement.

The main reason for the improved sound of the new Time Window, however, is almost certainly the further refinement of impulse response in the critical midrange and lower treble region. Pulses between 1 msec and 0.15 msec duration are now reproduced with even greater accuracy than before; the speaker has become virtually flawless in this range. Pulse form retention deteriorates rapidly with durations of 0.15 msec and shorter, possibly because of the tweeter problems observed. There's a trade-off here: better pulses from 1 to 0.15 msec, worse from 0.15 to 0.1 msec and shorter. And it seems to be a trade-off that favors the sound.

Overall, we find the improved Time Window to be the nearest thing to a high-fidelity speaker in a single package of moderate size and affordable price. The Tangent RS2, reviewed below, is even more transparent and uncolored from about 200 Hz on up and costs less, but its bass is unacceptable without the addition of a subwoofer. The Time Window, on the other hand, can be enjoyed as is, straight from the carton.

Now if they'd only fiddle some more with the enclosure tuning and that tweeter peak . . .

Hartley 24" Subwoofer

Hartley Products Corporation, 620 Island Road, Ramsey, NJ 07446. 24-inch Woofer-Driver, $375 each (without enclosure). Tested samples on loan from dealer.

This is widely accepted in audio extremist circles as the Mount Everest of subwoofers: when size, convenience or price is no object, it's supposed to be the summit of the art. Well, it isn't.

We tested it in a completely sealed and heavily braced custom enclosure, 4 feet wide by 2 feet deep by not quite 3 feet high, with an estimated internal volume of 18 cubic feet. It took four good men and true (admittedly not professional furniture movers) to lug a pair of these monsters from the van to the laboratory, and after the second one they swore they'd never do it again. In other words, the "infinite" baffling of the woofer was carried to the outermost limits of practicality, short of building it into the wall or constructing the enclosure in a room where it would have to stay forever. And, as it turns out, the Q of the woofer is much too high in a mere 18 cubic feet, suggesting that flat bass could only come out of it from the middle of an "infinite" wall.

We measured a 6 dB hump at approximately 48 Hz, indicating a system Q of 2. The -3 dB point of the system was at 33 Hz; the corner frequency as normalized for the hump was estimated to be 36 Hz. So the Godzilla of the low frequencies turned out to be a 36-Hz box, somewhere between the 10-inch Cizek and the 10-inch Snell—and not nearly as well damped as either. In fact, pulsing the woofer with four-cycle tone bursts at 100 Hz, 50 Hz, 40 Hz and 30 Hz revealed a distinct half cycle of ringing in each instance. The system was obviously underdamped, and a woofy, whompy quality on bass transients was easily ascertainable by ear. This is one area where measurements and the sound of music always correlate.

In all fairness to Hartley partisans, we must point out that despite its inability to reach into the 16 to 32 Hz octave and its inadequate damping, the system could move the air in the room like no other subwoofer known to us and was able to generate truly lifelike midbass pressure levels. Piston area isn't something to be sneezed at. Since the quantity of low-frequency energy has just as much to do with realistic sound reproduction as the quality, the Hartley remains in the running against more accurate but less potent subwoofers. But that's only because no one has bothered to design *mathematically* an 18-cubic-foot subwoofer of optimum performance. If the problem were thus stated and solved from scratch, the required driver parameters (moving mass, compliance, flux density, voice coil dimensions, etc., etc.) would be quite different from the 24-inch Hartley's.

Quite aside from the Q problems, though, there are some funny things about the basic concept of the driver design. Would you believe a 1½-inch voice coil moving a 21½-inch cone? It's like pushing a shopping cart with your pinkie. Would you believe a cone ribbed and stiffened for good *high-frequency* response? The damn thing goes out to 3 kHz! Who would want to use it above 100 Hz? And who wouldn't welcome some sort of smooth mechanical roll-off above a couple of hundred hertz to help make the crossover design a little simpler?

But that's not all. The 96-mm (3¾ in) aluminum tube that sticks out from the middle, purportedly to act as a heat sink for the voice coil, is alive! The axial response shows a double peak at 11.5 kHz only 10 dB below the bass reference level and a single peak at 29 kHz that's only 15 dB down. We don't think this high-frequency garbage could be activated through any reasonable crossover network, but we still wonder why the Hartley people threw in a peaky coaxial supertweeter at no extra charge.

One day, a nice, clean-cut boy who always did his physics and math homework and never consorted with long-haired, T-shirted audio mystics will graduate from engineering school. On the back of the envelope his diploma came in, he will design in one sitting a large but utterly simple subwoofer capable of producing profound and accurate bass while happily obeying the laws of nature. It will be the dawn of a new era, just like the day Galileo dropped those iron balls from the Leaning Tower of Pisa.

Innotech D24

Innotech, 42 Tiffany Place, Brooklyn, NY 11231. Model D24 floor-standing speaker system, $750 the pair. Five-year warranty. Tested samples on loan from manufacturer.

We can dispose of this review rather quickly, since the Innotech is directly competitive with the DCM 'Time Window' and doesn't sound nearly as accurate. Both are floor-standing units of the same height (3 feet), with comparable elbowroom requirements; both incorporate four drivers; both go down a little bumpily to 50 Hz and then roll off quite rapidly; both are intended to appeal to the enlightened audiophile; and both cost roughly the same (the Innotech is actually a bit more expensive). But the Time Window is unquestionably the more sophisticated and more highly perfected design.

The main weakness of the Innotech is the 1½-inch Mylar midrange dome, which rings severely throughout its range and interacts with the other drivers in various unwelcome ways, affecting both frequency and time response. With a better midrange driver the speaker might have turned out differently. We also question the extremely close spacing of the four drivers, which appears to be the cause of severe lobes in the polar response. The net result is a highly colored sound with a distinctly "canned" quality, the only redeeming feature being the absence of aggressiveness or "ear burn," probably because the tweeter is reasonably good.

A recent CBS Technology Center report on the Innotech D24, published in *High Fidelity*, contains the amazing assertion that "the tone-burst response is good." Did they check 2.5 kHz and 4 kHz, for example? And if that's good, what's bad?

Spendor BCI

Audio International, Inc., 1055 Thomas Jefferson Street NW, Washington, DC 20007. Spendor BCI vented-box loudspeaker, $570 the pair. Tested #13478 and #13479, on loan from distributor.

This nice-looking English box speaker, about two feet high by one foot square, comes with a good international audiophile reputation. We found it to be quite listenable; however, both the Rogers LS3/5A and Tangent RS2 are more refined examples of the same civilized BBC-monitorish sound and cost less to boot.

The Spendor BCI manages to achieve in a vented enclosure what we're already accustomed to in sealed-box speakers from England: completely underdamped bass. The composite response of woofer and vent shows a hump of approximately 6 dB, centering on 72 Hz. You can hear the hump, too; we found the bass boom decidedly objectionable regardless of speaker placement. From 100 Hz on up the response is very acceptable, especially at 45° off axis, where it remains within ±5 dB all the way to 20 kHz. The axial response is ragged, though, with crossover troughs evident.

In the time domain the speaker behaves better than most; pulse form retention is very good to excellent between 1 msec and 0.1 msec, but the woofer needs 2.5 msec to recover after a 0.5 msec pulse and the midrange driver keeps on going after a 0.1 msec pulse. The clear, focused, neutral sound of the BCI is probably due to the generally good impulse response; the midrange, however, sounds slightly rough.

When we switched to the Rogers after extended listening to the Spendor, the Rogers sounded considerably smoother and more transparent; as for the Tangent, we prefer it even to the Rogers and therefore unequivocally to the Spendor. All of which doesn't mean that the Spendor BCI isn't a rather nice speaker (except for that bass boom), but as we've said before, we don't give consolation prizes.

Tangent RS2

Sound Physics Labs, a division of Inception Audio Ltd., 2 Carlton Street, Suite 919, Toronto, Ont., Canada M5B 1J3. Tangent RS2 Reference Speaker, $480 the pair. Lifetime warranty (conditional). Tested #077102A and #077102B, on loan from importer.

The ink was hardly dry on our enthusiastic review of the Rogers LS3/5A when we received our samples of this somewhat larger but still highly portable and basically similar British import. And, would you believe it, the Tangent is an even better speaker. Rule, Britannia . . .

The Tangent RS2 is just one of a whole family of interesting speakers designed by John Greenbank in England, but it's the only one we've been able to get our hands on so far. (The only North American source we're aware of at this writing is the Canadian importer listed above.) Model RS2 is a completely sealed box of approximately 19 liters (2/3 cubic foot) internal volume, containing the famous KEF T27 dome tweeter, an Audax 8" woofer and a "phase-corrected" crossover network. The basic concept of the design is to achieve a reasonable degree of time alignment between the two drivers while mounting them close together in an ordinary rectangular box, i.e., without stringing them out on a staggered baffle. This is the main difference between the Rogers and the Tangent; the former can be time-aligned only by turning the woofer end closer to the listener than the tweeter end, whereas the Tangent is acceptably aligned when listened to head-on.

It's interesting to observe how differently these two speakers utilize the same tweeter. In the Rogers, the KEF T27 is covered by a perforated screen; in the Tangent it sits naked. As a result, the Rogers goes out comfortably to 18 kHz, but the Tangent is *flat* to 32 kHz and doesn't show signs of quitting up to almost 40 kHz! We really don't know the reason for this difference in application or whether it has anything to do with the audible differences between the two speakers—but what a tweeter! If it were more efficient and able to handle more power, it would undoubtedly be every speaker designer's automatic first choice.

As a full-range system, the Tangent is very close to dead flat in response (especially at its "sweet spot") from 300 or 400 Hz on up into the first ultrasonic octave. From about 200 Hz *down*, however, it's another story. Would you believe a hump of 8 dB at 75 Hz? It's that English bass again, grossly underdamped (the Q is about 2.5) and distinctly audible as a thick whomp that underlies the otherwise startlingly beautiful sound of the speaker. The only thing that could cure it would be more magnet, and more magnet would cost more pounds sterling. (We should never have given up India, sir.) Still, a very flat speaker overall.

In the time domain, the RS2 shines even more brightly. Pulse reproduction is truly excellent anywhere between 1 msec and 0.1 msec duration—and even a little beyond either end of that range. The angle at which pulse coherence is best is still somewhat off axis toward the woofer but far closer to a normal listening angle than in the case of the Rogers. On the whole, we haven't encountered a single electrodynamic speaker system that equals the Tangent in this area, since even the improved DCM Time Window falters on the shorter pulses. One would have to go to the $6000 Beveridge electrostatic to get better impulse response.

On the other hand, both the woofer and the tweeter of the RS2 exhibit some ringing at various frequencies when tested with tone bursts—not nearly as much as most speakers but certainly more than the improved Time Window and a bit more than the Rogers as well. The excellent pulse form retention (i.e., freedom from time smear) appears to carry the day, however, since the Tangent sounds more accurate than the other two—as long as you can mentally tune out that bass boom. (Pulling the speaker far away from all walls and raising it high above the floor will help considerably.)

When we say more accurate, we mean both smoother and clearer, better focused, more spacious and detailed, less colored. The Rogers sounds a little hard and zippy by comparison (only by comparison!) and the Time Window a bit less open (only by comparison!). The Tangent is the most electrostatic-sounding electrodynamic we've found so far.

As for adding a good subwoofer to eliminate that bass boom, it isn't as simple as it may seem. With an 8 dB hump at 75 Hz, you can't just insert a ready-made crossover, whether passive or electronic, at 100 Hz or even 150 Hz, and call it a day. The transition wouldn't be smooth enough. We're looking into this problem and may possibly come up with a recommendation in the next issue. (We're no longer experimenting with cross-overs or biamping for the Rogers, since we now prefer the Tangent.)

Meanwhile, we suggest that you use the Tangent RS2 as is (if you can get a pair) and try to forget about the bass. If you're a fanatic about clarity, as we are, you'll find the trade-off worthwhile. If not, you'll probably prefer the improved DCM Time Window, which is a better balanced full-range speaker and also considerably more efficient. (The Tangent needs a fairly powerful amplifier for program material with a high peak-to-average ratio; in fact, we were able to make the Mark Levinson ML-2 clip on it with a master tape of piano music.)

In any event, it appears that little boxes have come a long way.

Ultraphase 2501

Ultraphase, 2875 South Raritan Street, Englewood, CO 80110. Model 2501 floor-standing speaker system, $596 the pair. Wooden stands, $32 the pair. Ten-year warranty. Tested #10056-25 and #10057-25, on loan from manufacturer.

This somewhat offbeat three-way system is about the size of the largest bookshelf speakers but much heavier, with unusually thick and rigid walls, and designed to be placed off the floor on a low pedestal. The sloping front baffle of the sealed enclosure holds a staggered array of three drivers: 1" dome tweeter, 2½" dome midrange, and 8" woofer.

Of its type and its price range this is a very decent speaker, considerably more accurate in response than what seems to be generally available. It lacks, however, that extra measure of engineering refinement that gives speakers like the Rogers LS3/5A, Tangent RS2 and DCM 'Time Window' their special appeal to the serious audio enthusiast. It doesn't quite have their smooth, transparent, unstrained quality in the midrange and on top, and therefore isn't quite as musical and listenable.

Where the Ultraphase shines is the lower frequencies. For once there is no boom, no woof, no whomp. No hump whatsoever in the bass response. This is a low-Q design; in fact the response profile suggests a somewhat overdamped condition, which of course is still greatly preferable to the usual underdamped boom box. Taking the 200 Hz level as 0 dB, the -3 dB point is at 70 Hz and the -6 dB point at 45 Hz. This is the kind of smooth, gradual roll-off that can be judiciously re-equalized with a few dB of bass boost in the playback electronics while still preserving an acceptably low Q—a definite advantage in flexibility over the boom boxes. Needless to say (at least to those who believe that not even speaker designers can defy the laws of nature), the Ultraphase sounds considerably more natural and untroubled on heavy bass transients (plucked double bass, bass drum, left-hand chords on piano) than any speaker with a high-Q woofer. On the other hand, Ultraphase's claim that "you will feel frequencies below audibility . . . you will feel a concrete floor vibrate" is the most arrant nonsense. No 8" woofer, regardless of its small-signal response characteristics, has the large-signal capability to do that.

Above 200 Hz, the Ultraphase can be adjusted to be quite flat in amplitude response by fiddling with the midrange and tweeter level controls in the back. We doubt, however, that the consumer could do this accurately without laboratory instruments and we disagree with the entire philosophy behind such controls. (They can't really make the speaker response zig where the room characteristics zag—that takes infinitely more sophisticated equalization—and they destroy the impulse response of a speaker when incorrectly set.) We found the best setting to be quite far down (10 o'clock) on the "mid" control and all the way up (3 o'clock) on the "high" control. That still didn't make the speaker as flat as a Snell or a Canton or a Tangent, but it was good enough to obviate all complaints.

In the time domain there was good news and bad news. The Ultraphase is claimed to be time-aligned, and its drivers are staggered ostensibly to that purpose. Our tests indicated that the leading edges of the wave fronts from the three drivers coalesce only when the speaker is tilted much further backward than is practical, with the bottom side partly exposed to the listener and the woofer facing not far from straight up. Pulses of 0.45 msec duration were the shortest that the speaker was able to reproduce at any angle; narrowing the pulse toward 0.1 msec resulted in increasingly severe pulse form deformation. Pulses between 1 msec and 0.45 msec duration (probably more important from the listening point of view) were reproduced quite accurately.

The cavities that the thick front baffle creates below the midrange driver and the tweeter as a result of the staggered mounting arrangement (flush mounting on a stepped baffle would have been better) must take the blame for resonances and reflections that have exactly the same effect as ringing. Actual diaphragm ringing was observed only in the neighborhood of 5 kHz; above 7 kHz the reflections from the baffle cavities made accurate tone burst testing impossible. Overall, the midrange driver definitely appeared to be the least well damped. The woofer exhibited no ringing at all when driven moderately, but at very high levels its initially low Q (perhaps 0.4 or 0.5) rose to approximately 1.0, and there developed a moderate amount of ringing. This is to be expected in all cases where the number of voice-coil turns in the gap isn't absolutely constant regardless of excursion.

The net audible result of all this is very good balance, imaging, clarity and focus, but with a definite edginess that we found fatiguing. It seems that some of the ringing occurs at the most irritating frequencies. The bass, as we said before, is very classy, but of course it's still 8-inch bass.

One thing that detracts severely from our basically positive reaction to this thoughtfully conceived speaker is the promotional literature Ultraphase is using to sell it. For a small company, it looks as if quite a few dollars were spent on a 2501 brochure, a more elaborate brochure on the four previous Ultraphase speakers, an "Audio Consultant's Handbook" and so forth—all of them full of the most egregious misinformation. The worst of it is that the literature takes an extremely detailed, technically informative, almost educational approach, which is likely to mess up innocent heads very effectively and for a long time. Among the more flagrantly erroneous principles promulgated by Ultraphase are that a heavy woofer cone can't reproduce bass transients as accurately as a lighter cone . . . that correct time alignment means positioning the voice coils along a single vertical line . . . that an impedance peak in a speaker constitutes an undesirable load for the amplifier . . . that a vented enclosure can't have flat frequency response . . . we could go on and on. (We've already mentioned the bit about shaking a concrete floor with an 8-inch woofer.) If this stuff were in a national magazine ad, we'd give it a good roasting in *The Admonitor*; as it is, we won't go further than to wonder aloud where it all came from. We've spoken to Bill Kennedy, the engineer who designs the Ultraphase speakers, and he seems to know what he is talking about. Was an advertising copywriter with a pop-tech background foolishly given his head?

The Ultraphase is a good speaker with some flaws that can probably be ironed out. The last thing it needs at this stage of its evolution is a credibility problem created by its own promoters.

Recommendations

The only change here, as a result of this last batch of tests, is our preference of the Tangent RS2 over the Rogers LS3/5A. Our other two recommendations remain the same.

Best speaker system tested so far, regardless of price: Beveridge 'System 2SW' (with reservations about subwoofer and headroom—see review).

Best speaker system at a much lower price: Tangent RS2 (with strong reservation about bass—see review).

Best full-range system per dollar (without major trade-offs): DCM Time Window.

A Sampling of Headphones with Audiophile Aspirations

By the Staff of
The Audio Critic

We examine four electrostatics and two electrodynamics (the latter, as it turns out, mainly for comic relief).

Headphones are exactly like loudspeakers and, at the same time, totally different. The similarity lies in the electroacoustic transducer format: in each case, the output waveform of an amplifier is translated into the excursions of a diaphragm, and the same principles apply. The difference is in the coupling to the ear: loudspeakers actually try to imitate life by producing a sound pressure gradient in the listening room, whereas headphones bypass the natural listening environment and "mainline" the pressure field directly into the ear canal. As a result, loudspeakers tend to sound more natural, since real-life interfaces are maintained; on the other hand, headphones have a better chance to preserve waveform accuracy all the way to the eardrum, since the signal goes through fewer transformations and, besides, the smaller diaphragm is more precisely controllable.

The impression critical listeners generally form about headphone listening is that more sonic information is revealed ("I hear things I didn't know were on the record") but that the experience is ultimately fatiguing and unsatisfactory. We feel that a pair of first-rate electrostatic headphones, such as the Stax SR-X/Mark 3, is an invaluable "audio loupe"—a magnifier for examining what goes on in a particular piece of equipment when inserted into a known chain of components. We also find the uncanny clarity of such headphones exhilarating, but in the long run we can't derive much musical pleasure from their use. We prefer our music by total immersion rather than intravenously, not only because it feels more normal that way but also because low bass is perceived more through the body than through the ear canal. Nevertheless, no serious audio enthusiast should be without a pair of good headphones; just to be able to play at any loudness level without intruding on others is a good enough reason.

Incidentally, none of the above remarks apply to binaural sound reproduction; that's a very different ball game from stereo, to be explored in depth in a future issue.

How we measured them.

The frequency-domain and time-domain measurements we made to evaluate speaker systems (see our Number 4 issue, page 20) are equally applicable to headphones. The problem, of course, is coupling. How do we create the same interface between the headphone diaphragm and the measuring microphone as exists in actual use between the headphone diaphragm and the ear? The standard solution is an "artificial ear," which is an acoustic coupler made of metal and designed with a somewhat arbitrarily chosen "official" internal volume of 6 cm^3. Since your coupler and mine are made of flesh, blood and cartilage, and since the cubic capacity of yours isn't necessarily the same as of mine, we aren't too happy with this solution. It doesn't appear to simulate the on-the-head acoustic transfer conditions closely enough to yield results we'd be willing to accept as "official." As a matter of fact, pressing the headphones a little tighter to your ears or moving them a few millimeters will sufficiently alter the existing acoustic impedance matches to create a whole new set of frequency response characteristics.

We therefore decided to work *around* the problem, taking a whole series of iterative measurements, some with no coupling at all, others with loose coupling by hand, tight coupling by hand, probing with the microphone between the ear and the diaphragm, etc., etc. The composite picture that emerged, while perhaps not quite accurate quantitatively, provided reliable information about the basic frequency-domain behavior of the unit under test, such as smoothness or roughness, major dips or peaks, upward or downward slopes, and so forth. We're satisfied that our measurements reflected the realities of on-the-head performance, since the results correlated pretty well with what we heard. All of this applies, of course, only to frequency-response testing; the time domain can be investigated with nearfield readings of pulses and tone bursts, just as if the headphones were small loudspeakers.

How we listened to them.

The program material we used for our comparative listening tests consisted almost exclusively of direct 15-IPS copies of 30-IPS original master tapes, mostly recorded with only a single pair of microphones for the main pickup. The tape deck used was a highly modified Stellavox 'Stellamaster'; the amplifier in each case was the Bryston 4B (except that the two Stax models were also checked on an experimental tube amplifier especially designed for them by Julius Futterman, with the electrostatic elements driven directly off the output tube plates).

Headphone listening is a solitary experience, but we made sure that at each listening session there were at least two pairs of educated ears available to swap phones and compare notes; on occasion there were three. The findings reported below represent the unanimous conclusions of staff members and consultants; there were no major disagreements at any point during the tests, and minor disagreements were quickly resolved upon repeated listening to the same passage.

The most surprising discovery was not that electrostatic headphones are superior even to the best electrodynamics but that the latter aren't even close. Listening in quick succession to the Koss dynamic, for example, and either one of the Stax electrostatics elicited giggles from each auditioner. The difference was funny. Since the price differences aren't so staggering (especially in the case of the Stax SR-5), we don't see how the audio purist can be expected even to consider dynamic headphones for critical listening.

Fontek Minifon A-4

Specs Corp., 1169 E. Chess Drive, Foster City, CA 94404. Fontek Research Minifon A-4 electrostatic unit with C-4 coupler, $300. Tested sample on loan from distributor.

Fontek headphones are designed, we're told, by a Mr. Niwa who was formerly with Stax and involved in the development of the SR-X. The Minifon A-4 electrostatic is being sold as a Stax beater, which it isn't—not quite, anyway. But it's good enough to have given us the impression on first hearing that it just might be.

The A-4 has a bright, forward sound, with lots of presence—not the crude, zingy sort, just very detailed with a good, clean edge. That's probably why some people rate it number one. Compared to the Stax SR-X/Mark 3, however, it comes off as a wee bit edgy, aggressive and, at the same time, slightly woofy. It's not an altogether neutral reproducer. This is especially apparent on complex material with a rich harmonic structure, such as massed brasses.

The frequency response is quite flat, though under certain coupling conditions it appears that there's a loss of upper highs with respect to the level of lower highs. The SR-X is definitely smoother overall. In the time domain, the Fontek exhibited a moderate amount of ringing; the rise time was not quite as fast as that of the SR-X; furthermore, our suspicion of some frequency response irregularies received time-domain confirmation on square pulses. On the whole, good but not brilliant test bench performances.

Physically, the Minifon A-4 is mini indeed: very small and light, designed to be worn with very little pressure on the head, but not quite as beautifully finished as the SR-X/Mark 3. The coupler box that goes between the amplifier and the phones looks well made, although we found an unforgivable piece of loose wire rattling around inside our sample, apparently a paring from the transformer leads before they were soldered into place. The coupler will drive two pairs of headphones but does not incorporate a headphones/speakers switch.

We must add that we've heard rumors of unit-to-unit variations in Fontek products, although we certainly don't believe that the A-4 we tested was in any way defective. If there are better-sounding samples out there, they must be awfully good.

Koss 'Auditor' Dynamic/10

Koss Corporation, 4129 N. Port Washington Avenue, Milwaukee, WI 53212. Auditor Series Dynamic/10 stereo headphone, $85. One-year warranty. Tested #0495, on loan from manufacturer.

This is the top-of-the-line electrodynamic stereo model of the world's leading headphone manufacturer. As such, it comes off as a distinct disappointment. Compared to even the least expensive electrostatic we tested, it sounds like a real mess—jumbled, unfocused, not at all transparent, rough, and quite fatiguing. Also rather uncomfortable on the head. In all fairness, though, it must be pointed out that the competition in this group of six headphones was quite formidable; as one of our most experienced auditioners remarked about the Dynamic/10, after agreeing with its last-place ranking, "If you think this is bad, you ought to hear what *really* bad sounds like." True enough.

Our measurements confirmed the listening tests: the Dynamic/10 is totally incoherent in the time domain. Pulses of *any* duration are reproduced (if one can speak of reproduction in such a case) without the slightest resemblance between input and output. That almost certainly explains the lack of transparency and focus; the rough, irritating quality is probably explained by the ringing we observed on tone bursts. The frequency response also shows considerable roughness; between 2 kHz and 23 kHz there are jagged peaks and dips all over the place, and the bass rolls off at approximately 12 dB per octave below a 4 dB peak in the lower midrange (in the neighborhood of 300 Hz—measured with tight coupling). Not a nice response at all.

We don't quite understand why Koss is marketing the Dynamic/10 as an audiophile product. The Yamaha HP-1, for example, is a considerably better dynamic headphone and costs $20 less.

Koss 'Auditor' ESP/10

Koss Corporation, 4129 N. Port Washington Avenue, Milwaukee, WI 53212. Auditor Series ESP/10 electrostatic stereo headphone with E/10 energizer, $300. One-year warranty. Tested #0249, on loan from manufacturer.

As all good audio freaks know, the Koss ESP/9 was hailed as a reference standard by a number of reviewers and assorted audio professionals when it came out a few years ago. We have no opinion on the validity of that judgment, since we haven't tested the ESP/9; what we have here is the new model that Koss claims is a substantial improvement over all their previous electrostatics. The ESP/10 is both the most expensive and the most elaborate Koss headphone to date.

The top-of-the-line image is apparent from the massive E/10 energizer with its two VU meters, double outlets for two pairs of phones, selector button for phones/speakers, LED overload indicator, etc. Very impressive. More impressive, in fact, than the sound that results when these goodies are activated.

Not that the ESP/10 sounds bad; Koss couldn't get away with that for $300. But it unquestionably sounds the least accurate among the four electrostatics we tested. Specifically, transparency and definition of inner detail are well below the standard set by the other three; in addition, the aggressiveness we noted in the Fontek is also evident, perhaps even more of it. On the whole, a somewhat veiled, unfocused and not particularly pleasing sound, at least by comparison.

The frequency response of the ESP/10 is very flat at its "sweet spot," maybe the flattest of all the headphones tested in terms of maximum observable amplitude deviations, but the Stax SR-X/Mark 3 has a smoother curve. Under other conditions of measurement the Koss appears to have a very depressed midrange; which condition corresponds most closely to on-the-head use is hard to tell. In any event, a nasty standing wave exists at 11 kHz and another, not as obnoxious, at 20 kHz; we suspect the cavity behind the diaphragm to be the probable cause. Pulses were reproduced by the ESP/10 with slower rise time, more time smear and more ringing than by the SR-X; compared to the Fontek, however, the Koss wasn't significantly inferior. Thus the audible superiority of the Stax is well supported by the lab data; why the Fontek also sounds better isn't quite as clear, although certainly more explicable than the reverse would be.

One more thing. The ESP/10 is rather uncomfortable to wear, its earcup and headband design being identical to that of the Dynamic/10. At 3½ times the price, Koss could have done a little better. Overall, this ain't no reference standard.

Stax SR-X/Mark 3

American Audioport, Inc., 1407 N. Providence Road, Columbia, MO 65201. Stax SR-X/Mark 3 electrostatic "earspeaker" with SRD-7 adaptor, $230. Tested #02115 (adaptor #4062), owned by The Audio Critic.

This is without doubt the finest headphone we have heard and measured to date, but you must keep in mind that there are a number of contenders (such as, for example, the Infinity elecrostatic) that we haven't been able to test so far. Anything better than the SR-X/Mark 3 would have to be pretty sensational, though.

Phenomenal clarity, startling definition of detail, and freedom from colorations characterize the sound of these phones; the rendition of the input signal appears to be essentially complete, except of course for the low bass. In fact, there isn't much to say about the sound of the SR-X/Mark 3, since the usual anomalies that occasion instant comment just aren't there. Some people (not the ones we associate with) claim to detect some hardness or overbrightness at the higher frequencies, but we're virtually certain that what they're hearing is a merciless resolution of the intermodulation sidebands generated by their inadequately aligned phono cartridges.

The frequency response of the SR-X/Mark 3 is very flat and exceptionally smooth (the smoothest of all the headphones tested) out to 20 kHz and even beyond. Under certain conditions of acoustic loading there develops a large peak at 20 kHz; we could see it on the spectrum analyzer but couldn't hear it. In the time domain, this was the unit that rang the least, had the fastest time rise and created the least time smear among the six headphones we measured. The correspondence here between audible and measurable performance is as good as a reviewer who doesn't believe in magic could possibly ask of Mother Nature for corroboration.

The SRD-7 adaptor, which supplies the bias voltage for the electrostatic elements and provides the impedance match between the power amp and the phones, features outlets for two pairs of Stax electrostatics (any model) and convenient switching between loudspeakers and headphones (a red light comes on when the latter are energized). One thing that disturbed us somewhat was the introduction of 0.13 ohm additional speaker lead resistance per channel when the main speakers are switched through the SRD-7. We find that excessive, especially in cases where there are long speaker leads to begin with. For $230 Stax could have put in thicker wire to help keep damping factors high.

When we started **The Audio Critic**, the Stax SR-X/Mark 3 was already our reference headphone. After five issues' worth of listening tests, it still is.

Stax SR-5

American Audioport, Inc., 1407 N. Providence Road, Columbia, MO 65201. Stax SR-5 electrostatic "ear-speaker" with SRD-6 adaptor, $130. Tested #01228, on loan from owner.

For 56½ cents on the dollar, Stax will put you within a hairsbreadth of SR-X/Mark 3 ownership. We can't think of another "son of" product in the field of audio that approaches the performance of the high-priced "daddy" product as closely as the SR-5.

What do you give up when you buy the SR-5 instead of the SR-X/Mark 3? A tiny bit of smoothness and that ultimate refinement of inner textures—that's about all. The two models sound remarkably alike. In the laboratory, the only definite difference we could nail down was very slightly better pulse reproduction by the SR-X. On tone bursts, for example, both were equally excellent. We gave up after a while trying to find a major difference.

The SR-5 is slightly larger than the SR-X and, unlike the latter, completely surrounds the ear with its bulkier earcup. It's still very light, however, and comfortable to wear. Both models are open-backed and therefore not completely sealed against sounds from the outside.

Bargains are few and far between in equipment for the audio purist, but the Stax SR-5 is definitely one of them.

Yamaha HP-1

Yamaha International Corp., Audio Division, PO Box 6600, Buena Park, CA 90622. HP-1 Orthodynamic headphones, $65. Tested sample owned by The Audio Critic.

All we can say about this one is that it's a hell of a lot better than the Koss Dynamic/10, not to mention assorted nondescript electrodynamics, but it just hasn't got the focus and definition that seems to be the exclusive province of the electrostatics. It has excellent amplitude response, though, and you can hear it in the solidity, smoothness and balance of the sound: ±2 dB from 40 Hz to 1 kHz, ±3 dB from there on up to 10 kHz. The bass is particularly impressive; there's some roughness, however, in the top octave, with a big peak at 19 kHz preceded by a characteristic suck-out at 11 kHz. On the whole, flatter and smoother than most dynamics. Very listenable, too, without anything disturbingly unmusical to be singled out. Dynamic range is excellent.

It's in the time domain that things fall apart; the HP-1 is totally incapable of reproducing pulses of any duration. Furthermore, on tone bursts there's unmistakable evidence of ringing and time smear. Hence the lack of ultimate clarity and resolution of inner detail.

It would be unfair to dismiss the HP-1 without mentioning that it's extremely light, rugged, comfortable to wear and pleasant to use. Still, we can't wholeheartedly recommend it to the purist.

Recommendations

Even though six models constitute a very small sampling of what the headphone market offers today, there's a good chance that the recommendations below would have been the same if we had conducted a much broader survey. They represent the best of current thinking and execution.

Best headphone so far, regardless of price: Stax SR-X/Mark 3.

Close to the best at a much lower price: Stax SR-5.

A Comparative Survey of Power Amplifiers: Part III

By the Staff of
The Audio Critic

Our final installment (at least for a while) of this running survey is short but exceedingly sweet, as it presents both the best buy to date and the best, period. Plus some modestly sweeping generalizations about design criteria.

In the preambles to both Part I and Part II of this survey, we rashly held out the hope of finding more precise correlations between the measurable and audible characteristics of power amplifiers. We must confess that we didn't get nearly as far as we would have liked to, when suddenly and providentially help arrived in the form of an outstanding engineering paper by Eero Leinonen and Matti Otala, under the just-what-the-doctor-ordered title of "Correlation Audio Distortion Measurements" (*Journal of The Audio Engineering Society*, vol. 26, pp. 12-19). The paper accomplishes at least half of what we had in mind—and, we must hasten to add, better than we could possibly have done that half: it isolates five basic distortion mechanisms in amplifiers (three static and two dynamic types) and correlates five different distortion measurement methods (including several of ours) to evaluate the sensitivity of each method to each type of distortion. We couldn't have asked for anything more germane to our needs.

The other half of the problem—the one that the paper doesn't deal with—is just what these distortions sound like, singly and in combination, and just how much of each represents the threshold of audibility. Some insights along these lines may emerge from our future amplifier tests, which will be at least partly based on the Leinonen/Otala findings (published too late to influence this survey); furthermore, Dr. Otala and his team have already done some interesting research on the audibility of TIM, which will have been published by the time you read this. Andy Rappaport is also hard at work on sonic correlations in amplifier design (see his letter to the Editor in this issue), as are some good people at Mark Levinson and a few other places, so that things are definitely looking up.

Our new amplifier measurement procedures will go into effect beginning with the next issue and will be fully explained there;

meanwhile you may assume the same laboratory and listening tests to be behind the reviews below as were discussed in Parts I and II, except that the exquisitely transparent Tangent RS2 was added to the range of speakers used for listening and the GAS 'Sleeping Beauty' Shibata cartridge replaced the Denon as our reference cartridge in the few instances when we were not using master tapes.

What the best power amps have that others don't.

Our difficulties in fine-tuning the exact sonic correlations of our laboratory tests haven't prevented us from coming to some general conclusions regarding the specific circuit characteristics of good-sounding power amplifiers. It seems fairly obvious to us, after having measured and listened to all these different units, that the best ones have certain design principles in common.

Preeminent among these is *not too much negative feedback*. As we've explained before, zero feedback could be argued to represent the theoretical ideal, and a number of amplifier designers are now working on making it feasible. Whatever benefits negative feedback may provide (and no one can deny those benefits) fade into insignificance compared to the nastiness of excessive feedback. How much is excessive? Not very much. The "optimum maximum" can be mathematically derived for any given circuit and is generally in the 12 dB to 20 dB range. (The majority of amplifier designers appear to be innocent of these computations.) Thus a power amplifier with 50 or 60 dB of negative feedback is virtually guaranteed to be a screaming horror and unworthy of a serious reviewer's time and effort. (Look for ridiculously low *static* distortion figures at very high power, damping factors in the upper hundreds, and very low slew rates. Then start walking rapidly in the opposite direction.)

An inescapable prerequisite of not too much feedback, and therefore another feature of all good power amplifiers, is an *open loop of sufficient bandwidth and linearity*; in other words, the amplifier must be a pretty good one even before the feedback is applied by closing the loop. In the old vacuum-tube days, this was considered elementary; designers agonized over the performance of each individual stage of the circuit and then very gingerly applied a few dB of feedback to make everything a little better still. The solid-state revolution encouraged a cavalier attitude toward the open loop; any cruddy class B circuit could now be slapped together by a novice and then drenched in feedback as a cure-all, with high power output and magnificently low THD and IM figures to show for it on the spec sheet. The sound quality was another story, but who listens?

The remaining universal but frequently ignored criterion of a good power amplifier is a *stiff power supply*, capable of supplying the current required to make the amplifier a true voltage source regardless of the load, which in the real world may be resistive, capacitive or inductive—and most likely a combination of the three. Somehow the design of the power supply gets left out of the arguments among cultists about this kind of amplifier circuit versus that kind, but the incontrovertible fact is that no amplifier can be better than the stuff that comes down its power supply rails. What's more, the higher the maximum output-signal voltage the amplifier is intended for, the more critical the power supply. The Bryston 4B, for example, wouldn't be able to put 40 volts per channel into just about any load if it didn't have quite a bit of money sunk into its power supply. The somewhat skimpier RAM 512 begins to run into problems when you ask for 38 volts, whereas the Audionics CC-2 will happily give you 19 volts (and even a little more) into very nasty loads with its well-designed but relatively modest power supply. The *Electrocompaniet* amplifier, on the other hand, is in trouble with 14 volts at 4 ohms, even though it's a marvelous amplifier in many other ways.

A recipe for power amp design.

Singling out the above three criteria may seem a bit simplistic, since many other considerations obviously enter into the design of a good power amplifier. We haven't even touched upon such matters as protection circuits (current limiting is a potential source of sonic problems!) or the best method to deal with the very real distortion produced by out-of-band program components (not many designers are dealing with it at all, but we plan to delve into it in future issues). As we've pointed out before, however, **The Audio Critic** isn't an engineering school, and we must simplify matters, sometimes even flamboyantly, for the sake of effective consumer education. As long as our priorities are correctly ordered, we don't mind harping on the main issues one-sidedly or overemphatically. In that didactic spirit, we offer the following recipe for correct power amplifier design as the distilled experience of this survey:

(1) Make the amplifier as linear as possible before the application of feedback. (2) Apply feedback very sparingly and never mind the resulting lack of double-oh distortion figures. (3) Make the power supply as stiff as the budget permits—then spend a little extra and make it even stiffer.

Our very first review is a case in point.

Audionics CC-2

Audionics, Inc., Suite 160, 10950 SW 5th, Beaverton, OR 97005. CC-2 Amplifier, $399. Three-year warranty. Tested #04001, on loan from manufacturer.

This very new unit, representing a later stage in Audionics' thinking than the higher-powered and costlier PZ3-II, is the big surprise of this survey. It's the kind of amplifier the leading Japanese manufacturers have always advertised but never delivered: moderate in price and superb in sound quality. It seems that they know a few tricks in Oregon that are still unfamiliar in the land of Origami.

To our greatest amazement, the Audionics CC-2 blew away every power amplifier in our listening tests except the Mark Levinson ML-2. Does it sound better than the Bryston 4B? Yup. Does it sound better than the *Electrocompaniet*? Yes, indeed. Does it sound better than the Futterman? Well, we didn't have the very latest version of the H-3 available, but an earlier version didn't stomp the Bryston 4B and the *Electrocompaniet* the way the CC-2 did. It's enough to make the most seasoned audio purist walk away shaking his head and muttering to himself.

Let's specify what we mean by sounding better. The Audionics CC-2 equals in transparency and transient detail the Bryston/Electro level of performance and is at the same time smoother and sweeter on top. We never thought of either the Bryston or the Electro as hard or zippy—on the contrary, we found them far superior to others in that respect—but the CC-2 makes them appear that way. Make no mistake about it: we aren't talking about the thick, whipped-cream kind of smoothness without clarity. The CC-2 is both clear and smooth. Only the $1800-per-channel Mark Levinson ML-2 beats it on clarity, definition, inner detail, and total freedom from fuzz.

It mustn't be assumed, of course, that the Audionics can drive with ease every load the Bryston can. The 4B is capable of 40 volts per channel into any load down to 3 ohms or so; the CC-2 will deliver at least 19 volts, even into 2 ohms, and as much as 25 volts into 8 ohms. (Its official rating is 70/70 watts at 8 ohms, i.e., not quite 24/24 volts.) Thus it's far from a super amplifier in sheer output capability—but wait. There's a toggle switch in the back that bridges the left and right channels into a single mono amplifier rated at 225 watts into 8 ohms (that's more than 42 volts). What's more, the slew rate of 36 V/uS in the stereo mode is automatically doubled in the bridged mono mode. Now *that* begins to look like a super amplifier—and still for only $798 the pair. Unfortunately we didn't have two CC-2's available to listen to in the bridged mode as a stereo pair; we've been promised a second one for a follow-up report in the next issue.

Now comes the interesting part: why does the Audionics CC-2 sound so good? You've probably guessed it already. It's a perfectly simple, straightforward circuit with not too much negative feedback, so that it produces very little dynamic distortion. Its static distortions are far from spectacularly low; none of our THD, SMPTE-IM and CCIF-IM measurements were in the double-oh or even low single-oh range. High single-oh or point-one-something are typical at 20-plus volts out into a variety of impedances. Nothing to brag about on a Japanese spec sheet. Live and learn.

We're told that Audionics worked very closely with Fairchild, the maker of the output transistors, in developing the CC-2 circuit. That may explain part of its success; engineers who design solid-state devices generally know more about their application than circuit designers who look up the devices in a catalog. In any event, we're both impressed and delighted. Since the low-profile CC-2 matches the almost equally excellent BT-2 preamplifiers both in appearance and, within a few dollars, in price, Audionics appears to have a very good thing going for a large number of audiophiles.

GAS 'Grandson'

The Great American Sound Co., Inc., 20940 Lassen Street, Chatsworth, CA 91311. Model A-701M 'Grandson' Servo-Loop Amplifier, $329 (with meters). Five-year warranty; customer pays all freight. Tested #A-701356, on loan from manufacturer.

"Hard as nails," commented one of our auditioners, and we concur. This is not a nice-sounding amplifier. Very hard, yes, but also peculiar in balance, nasal and sizzly. "Thorns on the cymbals" was another comment. The listening sessions were short, since the sound was extremely fatiguing.

We can't even blame it on Jim Bongiorno, who had left the company before this bottom-of-the-line power amplifier was finalized, leaving only the fading mark of his funky nomenclature on it. (You know whose Grandson it is, don't you? Get the laser guns, professor . . .) Perhaps a genuinely accurate amplifier is too much to expect at this price, but we can't refrain from hoping, especially in view of the extraordinary quality of the Audionics CC-2 for not too many dollars more.

The Gas 'Grandson' is designed as an 18-volt amplifier (40/40 watts into 8 ohms, 80/80 watts into 4 ohms), and near its maximum output its distortion figures are far from impressive. If we still had our former faith in CCIF-IM, for example, it would be easy to say "Aha!" because the 13 kHz sideband alone is of the order of 0.2%, but the Leinonen/Otala findings tell us not to trust this excellent test in all cases and for all diagnoses. If forced to take a guess at the root of the sonic problem, we'd say it was the power supply, since the amplifier distorts several times more into 4 ohms than into 8 ohms, indicating insufficient current capability. We also noticed some peculiar cross-coupling effects between the two channels through the power supply, as well as an unhealthy amount of ringing with capacitive loads. So there seem to be enough things not to like about the Grandson on the test bench. We must admit, though, that we didn't dig very deep after those uninviting listening tests.

We hope that this model, or at least our sample of it, is untypical of the direction the GAS Company is now taking. Otherwise the successor to Bongiorno may turn out to be Addio.

Mark Levinson ML-2

Mark Levinson Audio Systems, 55 Circular Avenue, Hamden, CT 06514. ML-2 Class A Power Amplifier, $1800 (per mono chassis). Five-year warranty; customer pays all freight. Tested #1196 and #1199, on loan from manufacturer.

Here's the chance of a lifetime for the avenging consumerist: a 25-watt *mono* amplifier priced at $1800—and no discounts. The head of Louis XVI couldn't have been a more tempting object to Robespierre than this arrogantly elitist piece of hardware is to the long-suffering audio reviewer with the slightest taste for journalistic retribution.

Well, sorry to disappoint you, but in our considered opinion this is the best power amplifier in the world—at least the world we're aware of. Nothing we've ever heard equals its absolutely focused, pristinely delineated, rock-solid and totally unfuzzed sound. Nothing. Other amplifiers, even very good ones, that are A-B-ed against it invariably create the impression of slight distortions, smears and colorations. It's unfair but there ain't no Santa Claus; money talks and everybody else ends up with second best. (No one held a gun against your head to make you become an audio perfectionist, right?)

A word about that 25-watt rating. That's what the ML-2 will deliver into the official 8-ohm load, at any frequency. In other words, it puts out a little over 14 volts. But—a big but—it will put 14-plus volts into *any* load, including the kitchen sink or your big toe; for example, it will give you 100 watts rms into 2 ohms, which is just over 7 amperes rms or 10 amperes peak. And you can't call any amplifier with a 10-ampere peak capability a little amplifier. The ML-2 is a big amplifier (it weighs 65 pounds with its giant heat sinks and carrying handles front and back), but it's big on current rather than voltage. That's one reason why it sounds as good as it does. Of course, it will clip if you push it beyond 15 volts or so; we've done it. The solution is to bridge two of them; an inverting second input is provided for that very purpose and you've got yourself a 28-volt (i.e., 100-watt, at 8 ohms) amplifier for only $7200 the stereo pair. (Take their names, comrade. They'll be the first to be liquidated.)

The circuit of the ML-2 is the epitome of the 3-point design philosophy we discussed above: the open loop couldn't be more linear since it's pure, unqualified class A (in fact the amplifier is designed to maintain class A operation over its full output range even with a 2-ohm load—probably a world's record); the total amount of feedback used is extremely low; and the power supply is as stiff as you'll ever see. Quite frankly, the one thing we can criticize about the amplifier is that possibly inaudible design trade-offs were obviously not even investigated in its development; "if it's theoretically better, do it" was the guiding principle. Who knows, maybe an equally good-sounding unit could have been built for quite a few hundred dollars less. Certainly, at least $200 could have been chopped off the price if the front panel were less magnificently sculptured, but then a Rolls Royce owner wants that Rolls Royce grille up front.

Our measurements of the Mark Levinson ML-2 (admittedly BLO—Before Leinonen/Otala) could reveal no sins whatsoever. All the numbers were beautiful: THD, SMPTE-IM, CCIF-IM, propagation delay, recovery time, the works. If there's something wrong with this amplifier, we don't have the test for it yet. Oh yes, for slew rate collectors: it's of the order of 100 V/uS and double that in the bridged mode with two units.

We can't conclude this review before coming back to the sound of the ML-2, although it's hard to talk descriptively about something that aims to be totally neutral and actually succeeds. We feel that the ML-2 lets through all the sound that's fed into it; we heard it retaining tiny ambience details, for example, that got lost in the soup through other amplifiers. And, always, we must keep coming back to that complete lack of fuzz. Next to the ML-2, the sound of any other amplifier we know of has hair on it. Also, the ML-2 controls the bottom end of a marginally damped woofer better than anything else we've tried.

Enough. After all, if the amplifier were less good than it is, it would be an outrageous rip-off. That it's still a highly plausible purchase is a credit to both Mark Levinson and Tom Colangelo, the engineer behind the design.

Recommendations

Keep in mind that the following is based on our tests of only 18 amplifiers and our working knowledge of a couple of dozen others. That's not the total power amp universe; surprises may be forthcoming in individual tests reported in future issues.

Best power amplifier, regardless of price: Mark Levinson ML-2 (or bridged ML-2's for more power).

Second best power amplifier (and only incidentally the best sound per dollar): Audionics CC-2 (or bridged CC-2's for more power).

The Cartridge, Arm and Turntable Situation: Part II

The laws of physics and of geometry are resoundingly vindicated by a new generation of Japanese moving-coil cartridges; the systems approach carries the day in turntable design; and Swiss precision yields a reference tone arm.

Our discussion of lateral and vertical tracking geometry in Part I, with the accompanying alignment instructions, turned out to be the hottest button we've pressed so far. The response has been overwhelmingly positive; reports from those who performed the alignments range from simple affirmations of an audible improvement to delirious joy, with not a single dissenter in the lot. Of course, there are also those who flatly refuse to try the alignments because of a vested commercial or emotional interest in incorrect geometry; we're sure that Mother Nature feels deeply hurt by their snub.

Perhaps it should be reiterated to the tiny handful of practitioners who are still trying to argue with us about the theory behind the alignments that it's simply not negotiable; you might as well argue with the Pythagorean theorem. The 37-year-old mathematical analysis by Baerwald of the relationship between lateral tracking geometry and signal distortion is complete and impeccable, and the 15 to 17-year-old studies by Bauer, Cooper, and especially Woodward of the VTA (vertical tracking angle) problem are equally unexceptionable. (The man to watch in connection with the latest research on VTA is Jim White, a staff scientist at CBS Laboratories.) Against this long-standing background it's an incredibly silly suggestion that either Mitch Cotter or **The Audio Critic** made up these things out of whole cloth; why not accuse the Surgeon General of having invented lung cancer? Of course, as we point out elsewhere in this issue, neglect of the alignments creates a disastrous credibility gap when the resulting phono signal is used as reference material in listening tests.

A couple of follow-up notes on the alignments:

Many audio enthusiasts seem to be willing to pursue the lateral alignment to the nth degree, as it needs to be done only once, but then refuse to get involved in the vertical corrections from record to record because they're a pain in the posterior. Sorry, guys. No good. The lateral alignment alone is fine for mono but not for stereo. If that's all you've done so far, you haven't heard yet what we're talking about. Until the whole industry adopts a VTA standard (see our comments further below), the price of totally unsmeared sound is eternal messing with the height of the rear pivot or the height of the record.

A number of people also had trouble with the eyeball determination of 0° tracking error at the two null points. This is admittedly not easy with cartridges that aren't perfectly rectangular; we're currently in the process of preparing considerably more detailed and novice-oriented instructions on the entire alignment procedure, scheduled for publication in our next issue (Number 6). Meanwhile we wish to emphasize that *almost* correct alignment still sounds vastly superior to no alignment at all. It's the kind of thing that's worth doing even approximately, otherwise your cartridge and arm might be off by a mile. (This is not to be read as an endorsement of sloppiness. Right on the nose sounds best of all.)

Our approach to cartridge evaluation.

Since this is the first time that we're specifically reviewing phono cartridges, we must state right up front that we don't have a sonically correlatable laboratory measurement procedure for them yet, such as we use (and have full confidence in) for evaluating loudspeakers. The nearest thing to a valid cartridge test we've seen so far, at least on the face of it, is the pulse-train method developed by a JVC research team in Japan. This is a time-domain-oriented test that requires elaborate and (for us) excessively costly data processing equipment for analyzing the results; there's hope, however, that this type of instrumentation will become affordable in the near future. The usual hi-fi magazine tests with the standard CBS, Shure, RCA and other test records (frequency response, separation, 1 kHz square wave, IM distortion vs. peak velocity, etc.) show quite poor correlation with what our ears tell us, except of course when the cartridge response isn't even in the ball park. We're most reluctant to get involved in any of these tests, unless further investigation raises our level of confidence in one or another of them. We're currently making a survey of all available test records, including some little-known ones.

The fact is, in any event, that a pickup system designed with a low enough motional impedance to track today's records without immediately audible difficulties will automatically have the bandwidth and high out-of-band resonant frequency to make it look good on most standard steady state tests. In other words, the important differences among the better phono cartridges are not in the areas routinely measured.

Then where are those important differences? We believe they are in the ability of the stylus and the generator system to replicate the exact time relationships cut into the groove by the cutter head. In Part I we explained the time-related aspects of the stylus/groove interface and how even a 5-micron anomaly in that unforgiving microworld can create audible time smear. The electromechanical structures inherent in different pickup designs are prone to such slippage to very different degrees, and our experience has been that the audible performance of each design correlates quite neatly with the common-sense suitability of its structure to the preservation of time-domain integrity. The paradox, then, is that audible differences that are difficult or impossible to document with conventional laboratory tests are generally predictable by inspection! (Needless to say, just about anything in this world that can be heard can also be measured, but not necessarily with measurement techniques that are readily available.)

Inspection turned out to be the most reasonable and effective elimination procedure in our attempt to find the best possible phono cartridge for the audio purist. Remember that auditioning a series of cartridges is unbelievably time consuming when both the lateral and vertical alignments are meticulously performed (and, of course, any other way the listening tests would be a total waste of time). If we attempted to listen to every cartridge whose maker raises his hand and shouts that he has the best, we'd be doing nothing else all year. The only solution is to look for *prima facie* attributes of correct design, from which perspective the cartridge universe suddenly shrinks to reasonable proportions and the listening can begin.

Fortunately, price differences can be ignored in this screening process. We firmly believe that the right cartridge for a \$2000 home music system is the same as for a \$20,000 system. The most that can be saved by skimping on the cartridge is a couple of hundred dollars. It isn't worth it. Every serious audio enthusiast ought to have the world's best cartridge; it makes a bigger difference in the end result than any other component except the speaker and it's by far the least costly step toward perfection. The fact that hi-fi dealers use cheap cartridges as a promotional item ("tell ya what I'm gonna do—I'll throw in the cartridge free of charge") shouldn't influence anyone who cares enough about audio to be reading this publication.

The VTA mess.

One criterion that immediately eliminates a large percentage of cartridges from serious consideration as state-of-the-art devices is vertical tracking angle compatibility.

A groove cut with a VTA of, for instance, 17° should be played back with a VTA of 17°. If it's played back at 19° instead, it may sound perfectly fine to some people but it won't—it physically can't—sound as clear, focused, uncolored and noise-free as it would if played back at 17°. This we have verified beyond all reasonable doubt and demonstrated to numerous professionals. Our mail, as we've already stated, also confirms it.

Now, all of today's records are cut within a VTA range of 3° or so, between a little over 15° and a little over 18°. As long as the built-in angle of the cartridge is anywhere close to that range, it's possible to match the VTA of the playback to the VTA of the cut, either by raising or lowering the rear pivot of the arm, or by raising or lowering the height of the record on the platter, or in extreme cases by shimming the cartridge at an angle in the headshell. But when the built-in angle of the cartridge is, say, 30°—forget it. The heel of the cartridge body will dig into the record long before the compensation is sufficient.

What cartridges are designed with vertical angles so large that compensation is impossible? Any number of ADC, Sonus, Grado, and Shure models, for openers. Plus lesser-known makes by the bushel. We hope we're making it clear that we aren't talking about anything terribly subtle or elusive. As a matter of fact, *exact* VTA measurements are still a subject of debate and continuing study among researchers. No, what we're dealing with here is *gross* deviation from the 15-plus to 18-plus bracket. Even though tracking force and dynamic conditions have a decisive influence on the VTA assumed by the stylus during playback, in many cases it's quite apparent visually that there's something wrong, as soon as the cartridge is taken out of its box. For example, the unloaded stylus beam of the Dynavector 20B appears to be sticking out at a 45° angle. You'll never get to 18° from there unless you step on the cartridge (which may be what you'll feel like doing with it).

The only sensible way out of this mess is an official (meaning IEC and RIAA) recording standard defining a specific VTA with which all masters must be cut, within the smallest tolerance that can be realistically demanded. Once the relatively tight little world of mastering studios has a decent standard, pickup manufacturers will either go along with it or not—but the serious audiophile will be able to align his phono setup permanently. If, however, the manufacturers are allowed to get into the debate about the need for a standard, we predict there will never be one.

In any event, on this one count alone, we can say to a lot of fancy cartridges what the kids chant on the playground: out goes Y-O-U.

Other eliminative considerations.

An immediate disqualification for acceptance as a SOTA contender is often provided right in the manufacturer's spec sheet. After specifying the inductance and DC resistance of the cartridge (per channel), the manufacturer tells you to load it with such and such resistance (usually 47K ohms) plus a recommended number of picofarads of capacitance, which includes all cables combined with the input capacitance of the phono preamp circuit. A quick computation will often reveal that what the manufacturer is telling you in effect is that the output of his cartridge must pass through a high-Q resonant filter circuit before it becomes flat enough in amplitude to be listenable. Sometimes the resonant frequency of the circuit is well within the audio band and the Q is 2 or 3. Such a cartridge is, by definition, an electromechanical device of very poor time-domain integrity; it will store and release energy in various ways that aren't reflected by its amplitude response—and it can't possibly sound as good as a cartridge that requires no such inductive-capacitive equalization.

Quite frankly, we're uncomfortable with the inductances of all so-called magnetic cartridges except the Grado series (where the inductance is limited to 55 millihenries in all models). This is the root of the whole input-capacitance hysteria in preamplifier design. Moving-coil cartridges are of course inherently low in inductance and outside this controversy.

The rubber-tire suspension of the stylus cantilever in nearly all conventional magnetic cartridges is another source of time-domain trouble. These designs are prone to elastomeric stability problems; some technologists call them gym-shoe cartridges. Twenty-six years ago, Rabinow and Codier (the former is the "Rab" of Rabco) pointed out the existence of *needle drag distortion.* As the laws of physics haven't changed since then, it's still true that the stylus tip, in addition to its freedom to move laterally and vertically, also has an undesirable third degree of freedom that varies in extent according to the cartridge design: it tends to move longitudinally. (I.e, it pumps back and forth in the direction where the stylus beam is pointing.) This causes an unmistakable time modulation of the signal, and rubber-tire suspensions are particularly susceptible to it. What's more, the electrical generator configuration of a conventional magnetic cartridge is sensitive to this kind of mechanical excitation and will produce an output, whereas a moving-coil generator, for example, is quite insensitive. (See also Mitch Cotter's letter on page 23 of our Number 2 issue.) Here again, Grado cartridges have an advantage over other magnetics by using a stiff axial tieback (but Joe, you made the VTA too big!). It must also be pointed out that the built-in VTA value of a cartridge can shift around wildly as a result of gym-shoe compressions and deformations.

The moving-coil advantage.

When all is said and done, all inspections and screening desiderata sorted out, it would appear that the moving-coil designs are theoretically the most promising phono transducers. The theory, as we shall see, is borne out by our listening tests; let's first consider, however, the obvious advantages.

The moving-coil (MC) type of generator is electromagnetically more linear, being less subject to hysteresis, than magnetic generator systems in which the field moves rather than the coil. At the same time, the MC type is incomparably lower in inductance and therefore insensitive to capacitive loading. The MC configuration also possesses inherently better othogonality (i.e., true 45/45-degree response in stereo reproduction) and is inherently less sensitive to needle drag distortion, both by virtue of the stiff axial restraint the design lends itself to and because of the insensitivity of the generator to longitudinal forces. Perhaps most important of all, the MC principle permits inherently higher signal-to-noise ratios and therefore wider dynamic range.

This last advantage is frequently not understood by the typical audiophile, who figures that the higher the output of a cartridge the higher the achievable S/N ratio—and everybody knows that MC cartridges have low output. Wrong. MC cartridges have *high power output*—higher than conventional magnetics with equal stylus excitation—but it's achieved with high current rather than high voltage. Power in this case is measured as output voltage squared divided by the cartridge impedance, and that figure for the GAS 'Sleeping Beauty' for example is 15 nanowatts at the 1 kHz reference level of 5 cm/sec, whereas for the Shure V-15 Type III as a typical comparison it's 3.6 nanowatts. And it's the power (i.e., energy) output of the cartridge that determines the S/N ratio, not the voltage output. Of course, delivering the power to the preamp input requires proper impedance matching, which is what transformers (and head amps, for that matter) are all about.

With the vertical and lateral tracking geometry fanatically aligned in each case, we have proved to our satisfaction that *all* of the better moving-coil cartridges (and especially those of the most recent generation) provide greater transparency, clearer focus, higher resolution of inner detail, less coloration, wider dynamic range, and lower modulation noise than the best conventional magnetics. The sound is just more real, more lifelike. More about that in the individual reviews.

A word about stylus tip geometry.

Let's get this straight once and for all. A long and narrow contact area between the stylus tip and the groove results in the most accurate tracing, least distortion and lowest noise. It doesn't particularly matter whether this type of stylus is called Shibata, Pramanik, Stereohedron, Special Elliptical or whatever. It varies slightly from maker to maker, but it's the kind to get. Making the contact area less long or less narrow is a step backward.

We don't feel obligated to dwell on the scientific rationale behind this, since every serious researcher in the world is of the same opinion. The occasional dissent is strictly from audio-freak quarters. We can only manage a sad smile when we run into a spherical-tip cultist; we try to explain to him the need for alignment-before-judgment if he's willing to listen. (See also the comments on page 3 of our Number 3 issue.)

The listening setup.

Each of the cartridges reviewed below was first listened to in a Supex SL-4 lightweight headshell plugged into a Grace G-840F arm mounted on a Linn-Sondek LP12 turntable. All alignments and adjustments were carried out in accordance with the instructions set forth in Part I. Contenders for top choice were transferred to a Breuer Dynamic Type 5A arm mounted on a Thorens 126 Mk II turntable, our current reference system (not to be interpreted as a once-and-for-all endorsement). The cartridge in the latter system at any given moment became the reference (A) against which each newly tested cartridge (B) was judged.

The preamplifiers used were the Hegeman HPR/CU and Mark Levinson ML-1. Needless to say, only one preamp was used for both cartridges in any given A-B test. Moving-coil cartridges were played through the Verion Mark I transformer. The reference speaker/amplifier system was the Beveridge 2SW.

The arms and turntables also reviewed below were tested with the same precepts in mind as discussed in some detail in Part I. More finely tuned and revealing laboratory tests are now in the process of being evaluated; we have no more confidence in standard tests in this category than in the case of cartridges.

Breuer Dynamic 5A

Sumiko Incorporated, PO Box 5046, Berkeley, CA 94705. Breuer Dynamic Type 5A tone arm, $750. Fluid-damping option, Type 5C, $150. Tested #092, owned by The Audio Critic.

Since our wide-eyed preview of this bird of paradise in our Number 3 issue, we have bought one. If that makes us certifiably insane, so be it; to the rest of you crazies we must issue the warning that the price keeps rising with that of the Swiss franc against the declining dollar and that we therefore take no responsibility for the figures quoted above. What's more, the U.S.A. distributor isn't exactly passing the arm along without a profit. You've got to have religion to get involved in this one.

Imagine a Grace G-707 executed by Cartier—or maybe we should say by NASA—and you have a pretty good idea of what the Breuer Dynamic looks and feels like. Utterly simple, straightforward, functional, and unbelievably precise. The four-point gimbals suspension has no detectable play in the bearings—absolutely none—and at the same time the bearings seem to be totally frictionless. We've never seen anything like it. No time smear from *that* source! The straight, aluminum arm tube is both rigid and dead. So is the perforated, nonremovable headshell. Tracking force and antiskating are adjusted by turning precision knurled knobs. The arm height is very precisely adjustable during play, a great advantage to the VTA-conscious. (Although it's advisable to tighten the main setscrew once the desired height is found.) The cueing mechanism is smooth as silk. And, hold your hat, the lateral geometry is *almost* correct; a perfectly aligned cartridge ends up pointing *almost* straight along the axis of the headshell. On the other hand, the mounting instructions are wrong with respect to overhang (H.G. Baerwald, where were you when Erhard Breuer needed you?).

In fact, the rest of our minor quibbles with this gorgeous piece of equipment also have to do with mounting. For one thing, the arm is held to the turntable by means of a mounting collar that must be affixed to the board with self-tapping wood screws, a primitive solution that the Japanese abandoned long ago. The arm rest must be mounted by force fit into a snug hole—ugh! And there's no provision for adjusting the azimuth of the cartridge, the assumption being that the arm board is in a plane absolutely parallel to the top of the platter and that the mounting collar is absolutely perpendicular to the arm board—a lot to take for granted.

But the sound, friends, the sound . . . We hate to do this to you, but there's a slight improvement over any other arm we've ever tried, including a properly tickled Grace G-707. There's just more ambience information reproduced, more inner detail, less "hair" on the sound. You could say that the window that stretches between the speakers got an extra wipe. All this without the fluid-damping option, which we haven't been able to get our hands on yet. (The damping of the main low-frequency resonance of a cartridge/arm system is a whole can of worms that we plan to open fearlessly in a future issue; manufacturers' literature and audio-salon chitchat on the subject are hopelessly simplistic.)

The mass of the arm is low enough to work very well with modern cartridges having medium to high stylus compliance; whether the crazy-high compliances are successfully accommodated we don't know—and don't care. Our current reference cartridge, the GAS 'Sleeping Beauty' Shibata, is very happy in the Breuer. In general we feel that the exact value of the system resonant frequency as determined by the arm mass and the stylus compliance is secondary in importance (as long as it isn't ridiculously high or low) to the numerous other performance criteria we've discussed.

Should you rush out and buy a Breuer Dynamic? Yes, if money is no object and your system is otherwise exactly as you want it to be. No, if your system needs any other kind of upgrading. The difference between the Breuer and, say, the Grace G-707 isn't going to change your life. But there *is* a difference.

Denon DL-103D

American Audioport, Inc., 1407 N. Providence Road, Columbia, MO 65201. Denon DL-103D moving-coil cartridge, $267. Tested sample on loan from dealer.

This is the very latest of the outstanding MC's from Denon, one small but distinct step up in performance from the DL-103S that was until recently our reference cartridge. The cantilever has been changed and the motional impedance reduced, making the 103D perceptibly superior in transparency and definition of inner detail to the 103S. The very slight roughness or hardness on top that was the trade-off against the uncanny clarity of the 103S has also been ameliorated in the 103D; however, the GAS 'Sleeping Beauty' Shibata is smoother and sweeter than either of the Denons and even a little clearer, so that the superb 103D still isn't our overall top choice of the new generation.

But it's close, damn close, and if for some special reason you can get one quickly and cheaply, grab it. It's a great cartridge.

Dynavector 20B

Onlife Research, Inc., Tokyo, Japan. Distributed in the U.S.A. by Audioanalyst, Inc., PO Box 262, Brookfield, CT 06804. Dynavector 20B moving-coil cartridge (with beryllium cantilever), $219. Tested #700799, on loan from distributor.

Any moving-coil cartridge with sufficient output voltage to require no matching transformer or head amp is intriguing, so we had to try this one. Unfortunately its VTA is so impossibly large as to make even an approximate alignment hopeless, and incorrectly aligned its sound is unbearably steely and irritating. How the exact same structure would sound with the correct VTA we have no idea.

Some people claim to have tamed the steeliness by loading the cartridge down with a low-value resistor; we don't believe that the electrical coupling is tight enough in the 20B for successful use of resistive damping, but the issue is in any case academic in view of the VTA disqualification.

EMT Model XSD 15

Gotham Audio Corporation, 741 Washington Street, New York, NY 10014. EMT Model XSD 15 moving-coil cartridge, $420. Tested #3768-14, owned by The Audio Critic.

Before the arrival of the latest generation of Japanese moving-coil cartridges (GAS 'Sleeping Beauty' Shibata, Denon DL-103D, JVC MC-1), this should have been by all rights everyone's reference cartridge, except for two serious drawbacks. One of these is that the XSD 15 is already encapsulated in its own sealed headshell, ready to be plugged into a standard arm. That means it can't be twisted to optimize the lateral geometry of incorrectly offset arms. The other problem is that it's available only with a 15-micron spherical stylus, so that it has distinct limitations in tracing ability.

If it could be extricated from the headshell (some experimenters have done it) and equipped with a Shibata tip (nobody to our knowledge has done it), it might even give the newer MC's with their lower motional impedances a run for their money, as it happens to be a superbly designed phono transducer with tremendous dynamic range capability. Even as is, trimmed in to the best of our ability in an arm of near-correct geometry, it has a big, juicy, startlingly "present" sound with excellent bass—but with obvious high-frequency problems. You just can't get into those sharp corners with a fat ball-point.

We believe that EMT could make the best phono cartridge in the world if they weren't locked into this hidebound, reactionary format (probably purely for marketing reasons). These people know their basics. They're one of the two or three cartridge manufacturers, for example, to talk about frequency intermodulation (FIM) distortion (they've even put it in their spec sheet with a percentage figure!), proving that they're well aware of the time-dispersive category of phono distortions. So you can bet that they didn't mess up the VTA, either; it's specified as 15° and we found nothing to contradict that spec.

But—in cartridges as in other audio equipment—SOTA means getting your whole act together, and that EMT hasn't done.

GAS 'Sleeping Beauty' Shibata

The Great American Sound Co., Inc., 20940 Lassen Street, Chatsworth, CA 91311. 'Sleeping Beauty'/ Shibata moving-coil cartridge, $240. Tested #7111002, on loan from manufacturer.

In Japan, the 'Sleeping Beauty' series of MC cartridges is known as the Coral 777 series, of which this is the latest and most sophisticated. Outwardly it looks exactly like the 'Sleeping Beauty' Super Elliptical (Coral 777EX), which is $40 cheaper, except that the box has a pink label and the "gland" from which the stylus protrudes is also pink. Internally, we're told, the GAS Shibata model is different from all its predecessors, with a beryllium copper spring suspension and, of course, a Shibata tip on the stylus. The stylus beam doesn't appear to be any smaller than that of the Super Elliptical, which is quite tiny.

Whatever its antecedents, this is the best phono cartridge known to us and the one we ended up with in our reference system. Through the Verion Mark I transformer (with P strapping), its sound is so transparent, focused, detailed and free from background noise that it's scary. And there isn't a trace of roughness or hardness on top. We find this combination of clarity and silkiness unarguably *right*. What you hear is essentially the tape from which the record was cut. Of course, these judgments are predicated on meticulous alignment of both lateral and vertical tracking geometry.

The only cartridge we're aware of that may eventually surpass the 'Sleeping Beauty' Shibata is the JVC MC-1 (see review below) *if* JVC recognizes the need to fix that top end and does so with complete success. Meanwhile, this is The One—and who says we have a grudge against GAS?

Grace G-840F

Sumiko Incorporated, PO Box 5046, Berkeley, CA 94705. Grace 'Black Beauty' G-840FB tone arm, $145. One-year warranty; customer pays all freight. Tested sample on loan from manufacturer.

This is basically the same design as our special favorite among the more reasonably priced arms, the Grace G-707. The back of the G-840 is virtually indistinguishable from that of the G-707, including the excellent gimbals suspension; only the arm tube is heavier and is slightly curved to accommodate a removable headshell.

Except for its somewhat higher mass, the G-840 performs like the G-707, and of course changing cartridges in it is a whole lot more convenient. Don't use it with crazy-compliant cartridges, though; the higher mass will make the system resonance too low and you'll have an unstable setup on your hands, with the stylus pumping all over the place.

Other than that, the same very high recommendations apply as in the case of the G-707.

Grado Signature Model II

Joseph Grado Signature Products, 4614 Seventh Avenue, Brooklyn, NY 11220. Signature Model II stereo/CD-4 cartridge, $500. One-year warranty; customer pays all freight. Tested #1341, on loan from manufacturer.

A $500 phono cartridge? What on earth is in it in materials and labor?

Joe Grado, when confronted with that kind of inquiry, is wont to reply: "I'm not selling materials and labor. I'm selling knowledge."

A charming bit of Sicilian braggadocio, to which our Hungarian retort was: "It's either the world's best cartridge, Joe, or it's overpriced. There's no third possibility."

Admittedly, any Grado cartridge starts with several advantages over other straight magnetic (i.e., non-MC) designs. The 55 millihenry inductance (the same in all Grado models) makes preamp input capacitance irrelevant and, in conjunction with fairly high voltage output (2.7 mV in the case of the Signature II), assures very decent power output, comparable to that of a good MC in the midrange, though not at the higher frequencies. Furthermore, the stylus-beam tieback used by Grado is much less susceptible to needle drag distortion than the "gym-shoe" designs. And when it comes to sheer mechanics, ex-watchmaker Grado is the acknowledged master. Thus an all-out Grado cartridge, which the Signature II obviously is, should be the ultimate magnetic—which the Signature II probably is, except . . .

First the good news. What's special about the Signature II is that the motional impedance has been reduced to an unbelievably low value, resulting in stupendous bandwidth and a high-frequency resonance so far up in the ultrasonic range as to be of no consequence whatsoever. This cartridge is *fast*! It reproduces the highs with a remarkably effortless quality and absolutely no grain, no grit, no shatter. Truly, it's like silk.

The trouble is—it's like *black* silk. Smooth but opaque. The remarkable see-through quality of the best moving-coil cartridges, that almost palpable presence of ambience details and inner textures, of separate musicians in a solid space, just isn't there. Compared to the GAS 'Sleeping Beauty' Shibata, for example, this difference in clarity is quite apparent and takes no "golden ear" to appreciate.

Which brings us to two important conclusions. One is that there must be a whole school of audio designers (and reviewers, for that matter) who bring out the champagne and celebrate when that silky quality without a trace of unpleasantness has been achieved. They believe that's the end of the line. Why do they stop there? That's our second conclusion: because they've never lived with the real thing. Yes, they're familiar with the real thing *live* (Joe Grado is an operatic tenor of international class and certainly knows what live music sounds like) but they automatically discount it as an unattainable ideal. They've never been exposed at sufficient length to the real thing in a living room—say, a laterally and vertically aligned top-notch moving coil with something like the Beveridge at the other end. Because if they listened to *that*, they'd *know* and wouldn't be satisfied with black-silk reproduction. (Cf. our comments on the Pyramid 'Metronome' in our last issue.)

Why isn't the Signature II more transparent? First of all, there's the VTA problem. The angle is much too large. We went as far as we could to correct it, shimming the cartridge with its nose up until its tail was practically dragging on the record, but we aren't sure whether we managed to bring it in line 100%. It sounded a lot better that way, though, than with its top parallel to the record. Then there's Grado's peculiar double-ball-point stylus tip, which he considers superior to the Shibata type with its long and narrow contact area; what geometrical theory is behind this preference escapes us. Lastly, we're dealing here with the inherent electromechanical limitations of a moving-field type of generator, especially with respect to freedom from time-dispersive distortions. Maybe this is as good as a magnetic cartridge will ever get (VTA and stylus-tip problems aside).

To get back to where we came in: the

Grado Signature II *is* overpriced, since it isn't the world's best cartridge. But it's a very good overpriced cartridge indeed.

JVC MC-1

JVC America Company, Division of US JVC Corp., 58-75 Queens Midtown Expressway, Maspeth, NY 11378. MC-1 direct-couple type moving-coil cartridge, price NA. Tested #07300017 (manufacturer's advance sample).

This came to us unexpectedly and very late in the course of our tests, so we're far from through with it yet, especially in view of its extraordinary promise. In some ways we find it the most logical and uncompromising solution to MC design we've seen, but we're hoping for a fully debugged version, which our advance sample quite possibly wasn't.

You must understand that JVC, completely aside from their commercialism in the medium-fi mass market, is an important center of basic research in phono technology. Some of the world's best minds on the subject work in their laboratories, and this new and different MC is a reflection of their latest thinking.

The coil of the JVC MC-1 is an almost microscopic, flat, chip-like affair that intersects the stylus beam only about 1.75 mm behind the diamond, considerably closer to the deflected end of the cantilever than to the pivot. JVC calls the design direct-coupled, since the stylus tip and the coil move as virtually one and the same structure, and the entire electromechanical configuration of the cartridge has some distinct theoretical advantages, all related to the reduction of time-dispersive distortions. The audible result appears to bear out the theory.

Never have we heard midrange clarity like this. The sonic intimacy and totally focused inner detail of the MC-1 are startling, surpassing in that respect even the GAS 'Sleeping Beauty' Shibata. This is SOTA performance—but there's a great big fly in the ointment. The top end has a very irritating coloration, probably caused by a peak that may or may not be amenable to damping as the design undergoes refinement. The machine-run curve that came with our sample actually showed a 4.5 dB peak at approximately 18 kHz; since we don't put much faith in test records and in amplitude response testing in general, and since the cartridge came in as late as it did, we don't feel ready to comment on JVC's apparent nonchalance about this.

Meanwhile, without recommending the JVC MC-1 as an immediate purchase, we strongly suggest that you check it out and listen to it if you have a chance. In our opinion, it's the wave of the future.

Linn-Sondek LP12

Audiophile Systems, 5750 Rymark Court, Indianapolis, IN 46250. Linn-Sondek LP12 transcription turntable, $549 (new price). Two-year warranty on mechanical components, one-year warranty on electrical components; customer pays all freight. Tested #016397, on loan from distributor.

Let's state at the outset what needs to be stated: The Linn-Sondek is a very Spartan, stripped-down, strictly utilitarian turntable of outstanding, but not unique, sonic performance, promoted both by its maker (Linn Products in Scotland) and by its U.S.A. distributor as though it were possessed of transcendental technology that puts it in a class by itself and justifies its otherwise incomprehensibly high price. (Even in Britain it sells for around two hundred pounds sterling; in the U.S.A. its price has just gone up $90.)

The LP12 is able to operate at 33-1/3 RPM only (you can't play on it those fantastic 45's from Mark Levinson and Japanese RCA, for example); it has no stroboscope and no speed adjustment (our sample tested out just a hair too slow); it has so little torque that you can't wipe your records on it; and its construction details, except for the unquestionably precision-made belt-drive mechanism and platter, are cheap and flimsy.

That said, we must admit that the Linn-Sondek introduces no disturbances whatsoever of the stylus/groove interface in normal playback; the quality of reproduction obtainable with it is limited essentially by the cartridge/arm combination rather than by the turntable itself.

Why? Not because of any kind of fanatical perfectionism in its execution, nor because belt drive is the cat's meow, but because the LP12 is designed as a *system*. Those Japanese direct-drive turntables that the Linn-Sondek is consistently beating in promotional A-B demonstrations are more or less nailed to their bases without any serious attempt at suspension design. The LP12, on the other hand, uses the tried and true method (introduced by Ed Villchur of Acoustic Research many, many years ago) of clamping the motor and suspending the rest of the system separately on soft springs, making particularly sure that the platter and the arm board bobble up and down in unison, as a single unit. This provides excellent isolation of the stylus/groove interface from both mechanical and airborne feedback, as well as from vibrations originating in the motor. Thus the sonic superiority of the Linn-Sondek isn't due to some mystical advantage of belt drive over direct drive but rather to a simple system of isolation to which belt drive happens to lend itself easily and cheaply. For example, the Thorens TD 126 Mk II, which uses the same system of isolation, sounds every bit as good (yes, we've A-B-ed them) and is in other ways an incomparably more sophisticated and better finished turntable for about the same price. This is not to belittle the Linn-Sondek's beautifully made moving parts—but just remove its cheap fiberboard bottom cover and see how money is being saved on everything else under that chassis.

This general flimsiness has more than cosmetic significance; for example, the Linn-Sondek is quite sensitive to subsonic excitation (heavy footsteps, truck in the driveway, etc.) because it's too light in construction, so that its overall mass is small enough to start oscillating at the fundamental suspension resonance with very little energy input. Furthermore, that fundamental resonance is a bit high in frequency (we'd say 6 or 7 Hz instead of 3 or 4, as it ought to be) and not too well damped (i.e., high Q). That means tone arms will be dancing out of the groove if their fundamental resonance with a particular cartridge is in that same vicinity (instead of at 13 Hz or so where it should ideally be). There's no need for a turntable to be nearly as hot and live subsonically as the LP 12, although that won't affect its sound when you're listening to it sitting still (unless your subwoofer goes down to 6 Hz).

Our conclusion: If you already own a Linn-Sondek LP12, you won't improve the sound of your system by switching to another turntable, but if you're starting from scratch, we see no reason to buy one in preference, for example, to the Thorens TD 126 Mk II.

RAM 9210SG

RAM Audio Systems, Inc., 17 Jansen Street, Danbury, CT 06810. RAM 9210SG Semiconductor Phono Transducer System, $299. Tested #2215-270, on loan from manufacturer.

This is a strain-gauge pickup system, a whole different breed from phono transducers based on magnetic fields, whether moving or stationary. At our present level of understanding, we don't quite see how any strain-gauge design could possibly have the dynamic range of the best moving coils, nor are we particularly happy with the existing mechanical (i.e., nonelectrical) method of internally RIAA-equalizing the inherent amplitude-sensitive response characteristic of strain-gauge devices.

The reason why we haven't gone into the whole question more deeply in our general discussion of cartridges is that, shortly after the conclusion of this group of tests and not in time for review in this issue, we received a sample of what is unquestionably an important breakthrough in strain-gauge pickup technology, namely the latest version of the Win Laboratories SDT-10 Type 11. Regardless of how we end up ranking this unit against the best MC's, it's obvious even on superficial examination that it obsoletes all previous strain-gauge designs. We want to study it carefully.

The RAM 9210SG unit, which some people were proclaiming a number of months ago as a SOTA contender (hence its inclusion here), is based on the old and widely wholesale-dumped Matsushita (Panasonic) EPC-451C strain-gauge cartridge, with electronics by RAM. Dr. Win's new transducer element is so far ahead of the Panasonic in all basic parameters that no amount of electronic wizardry by Dick Majestic can wring comparable performance from the latter.

So, rather than to tell you why we aren't particularly happy with the sound of this RAM system quite regardless of what else is available, we'll simplify matters by noting its "nofault" obsolescence in view of the far more advanced Win at the same price. A full review of the latter is scheduled for our next issue.

Thorens TD 126 Mk IIB

Elpa Marketing Industries, Inc., Thorens and Atlantic Avenues, New Hyde Park, NY 11040. Thorens TD 126 Mk IIB Electronic turntable, $500 (without tone arm). One-year warranty; customer pays all freight. Tested #21749, on loan from distributor.

This excellent Swiss belt-drive unit is currently our reference turntable, not because we think it's the best in the world (we don't know yet what is), but because we haven't found anything equally convenient and flexible that performs nearly as well. (All our comments here apply to the naked turntable; we haven't been able to get our hands on the Thorens Isotrack arm that comes with it in the C version.)

The decisive factor in choosing this turntable in preference to all others known to us is its superb isolation. The method of suspension is essentially the old AR one, also used in the Linn-Sondek (see review above), but the fundamental resonance of the system is a few hertz lower in frequency, and also lower in Q, than that of the Linn. That, combined with a lot more mass, makes the system considerably more impervious to subsonic quaking, and the isolation from mechanical and airborne feedback in the audio range is every bit as good, if not better. So are the sonic results. The TD 126 Mk II appears to do nothing to the stylus/groove interface and therefore nothing to the sound. Compare it with something like the Luxman PD-121 and you'll be amazed at the difference good isolation can make.

It's also interesting how much more turntable Thorens gives you for your money than Linn. The TD 126 Mk II has a very similar belt drive, but the motor is electronically regulated, so that speed changes are no problem at all. Push buttons select 33 1/3, 45 or 78 RPM (when did you last see a *three*-speed turntable of audio purist caliber?) and fine-control of each speed is possible via a knurled knob and illuminated stroboscope. Torque there isn't much of (though more than in the case of the Linn-Sondek), but that's belt drive for you. The details, finish and general feel of the turntable are very nice. On the other hand, it could be argued that the platter and drive mechanism of the Linn are a little more precisely made; the audible end result, however, is the same. And you don't have to tiptoe around the Thorens, even when the cartridge/arm resonance happens to be too low. It's an easy and pleasant piece of equipment to use; even changing the tone-arm board is a piece of cake.

And where would you rather order a piece of cake, Switzerland or Scotland?

Recommendations

The following choices reflect the findings of both Part I and Part II. Remember, though, that there's more coming in the next few issues.

Best phono cartridge tested so far, regardless of price: GAS 'Sleeping Beauty' Shibata.

Best cartridge per dollar: forget it (you can afford the best—at least in cartridges).

Best tone arm tested so far, regardless of price: Breuer Dynamic Type 5A.

Close to the best at a much lower price: Grace G-707.

Best turntable tested so far, regardless of price: Thorens TD 126 Mk IIB.

Best turntable per dollar: Kenwood KD-500.

Preamplifiers Revisited (No, Not Again!)

By the Staff of
The Audio Critic

Yes, again. Because the situation has changed considerably, and for the better. The unlistenable horrors are slowly fading into the past, close-to-SOTA performance has become very affordable (would you believe $379?), and the best has gone right through the roof.

Everybody is making a preamplifier these days; it seems to be the in thing to do, or at least the preferred way of obtaining entree into the audiophile marketplace. The number of preamp-only manufacturers, and of those who started out that way, is legion. The consumer is, of course, the beneficiary; good preamps are becoming easier to find than bad ones, and the price of the good stuff appears to be trending counter to inflation even if not exactly plummeting.

Comparing this latest batch of preamps to the ones covered in our Number 1 and Number 2 issues, we discern a distinct improvement; even the worst of them is considerably more listenable than, say, the BGW 202 (now apparently off the market).

We're still having considerable difficulties trying to correlate our laboratory measurements of preamplifiers with what we actually hear coming out of them through our reference system. The new measurement procedures we're about to phase in for our power-amp evaluations will be at least partly applicable to low-level amplification stages as well (see the power amplifier article in this issue, so that we won't have to repeat ourselves needlessly here); we expect these new methods to give us a better understanding of the measurability/audibility relationship in preamp distortions, but they aren't reflected yet in the reviews below. Even so, the tests reported here differed to some extent from the earlier series, owing to certain refinements of our usual measurements and listening procedures, so that a few comments are in order.

The RIAA equalization mess.

We now have instrumentation for measuring the RIAA equalization of preamplifiers very quickly, completely and accurately. (It used to be a slow and painful chore.) As a result, we've become aware of some interesting anomalies, goofs and impasses.

First of all, there's complete anarchy among preamps at the top end of the audio range. If the 20 kHz response is to be exactly -19.6 dB relative to 0 dB at 1 kHz, as the official playback characteristic dictates, then the 6 dB per octave slope must be continued to a very high frequency in the equalization network. If the circuit designer finds it necessary to stop the roll-off with a "corner" at, for example, 50 kHz, there will be an error of a fraction of a dB at 20 kHz and at even 15 kHz. Such corners are put in at different frequencies by different preamp designers; furthermore, they're also put in at different frequencies at the cutter end, in the recording preemphasis curve. (We're told that the Neumann and Ortofon cutter head people have finally gotten together on this and will henceforth provide identically leveled-off preemphasis curves, but what about your present record collection?) We strongly suspect that worst-case mismatches in this preemphasis/roll-off process will be audible on fine-tuned systems.

The solution would be an emendation of the standard; in addition to the specified time constants of 3180, 318 and 75 microseconds, a fourth one (say, 3.18 microseconds) could be included, so that the high-frequency slope doesn't by implication extend to infinity. But what do the standard committees come up with instead? Nothing on the high-frequency end—but a needless and unproductive complication on the low-frequency end.

A new IEC Recommendation, which has not yet been made a standard by the RIAA but is already being used by a number of preamp manufacturers (for example, Audionics and Hafler), adds a 6 dB per octave bottom-end roll-off to the existing playback curve, with the -3 dB inflection point at 20 Hz. This is ostensibly to keep undesirable low-frequency energy (from record warps, turntable rumble, arm resonances, etc.) out of the power amplifier and speaker, but it happens to be more effective as an obligatory tone control for bass cut than as a subsonic filter. At 5 Hz, the low-frequency garbage (if it's there) is attenuated only by 12 dB, which isn't enough; on the other hand, the response is already down 1 dB at 40 Hz, which is audible on a system with good bass response. Since standard committees are hard to argue with after the fact, the best way out under the circumstances would be to make this extra RC roll-off available via a separate two-way switch (marked RIAA old/RIAA new), so that the preamp would be officially up-to-date but still not irretrievably frozen in the new format. One thing is certain: no record will be cut with the inverse of this new playback curve, as it would require ridiculously wide groove spacing.

Another widespread RIAA boo-boo is in the equalization network used for creating the required time constants. Too many circuit designers choose a topology that can't possibly work accurately, even with resistors and capacitors of the tightest available tolerance. The circuit elements interact in such a way as to produce more than just the intended inflection points; there are additional little bulges and saddles in the response that simply can't be trimmed out, only moved from one place to another. The likelihood of audible colorations depends on the location and amplitude of these bulges and saddles. The whole problem is predictable and avoidable but too often not predicted and avoided.

Listening test conditions.

As we stated the first time the subject came up, the only justifiable purpose of an expensive preamplifier is to reproduce phonograph records accurately. (If all you need is a source selector switch and a volume control, there are simpler and less costly solutions.) Therefore the only way to evaluate the listening quality of a preamp is to play records through it—with the best possible cartridge and arm.

Since moving-coil cartridges are far ahead of conventional magnetics in bandwidth, slewing speed, dynamic range and freedom from time smear, we're now convinced that the only way to wring out the phono stage of a preamp to the limit of its capability is to drive it with the best MC one can find. In our case that was the GAS 'Sleeping Beauty' Shibata (mounted in the Breuer Dynamic 5A tone arm on the Thorens TD 126 Mk IIB turntable—see the reviews in this issue). The impedance match to the phono input was effected via the Verion Mark I transformer, the only completely neutral, no-sound-of-its-own device for that purpose known to us. The approximately 80 kHz bandwidth of the Verion allowed most of the out-of-band program components coming out of the cartridge terminals to get through to the first stage of the phono circuit and separate the preamps that could handle that kind of input from those that couldn't. Remember, even with perfect lateral and vertical alignment, and even with an optimally tracing Shibata tip, there's more than just audio coming out of those grooves; a fast and accurate transducer such as the 'Sleeping Beauty' will generate fairly high-amplitude out-of-band signals, which will either pass through without a hitch or excite in-band IM products, depending on the phono circuit design.

This is one of the important things that Mark Davis ignored in his experiments "proving" that all preamps sound alike; the Shure M91E he used is a slow, bandwidth-limited device that makes very light demands on the phono input stage. You might as well test-drive an automobile at 30 miles an hour and then decide that it handles just like all others.

For the same reason, we mistrust those straight-wire bypass tests of preamplifiers that use a tape recorder through an inverse RIAA network as the program source. Quite aside from the fact that it's never 30-IPS wide-track master tape that's used (so as to get some bandwidth and dynamic range in there, comparable to what you can get off a direct-to-disc recording with an MC cartridge), these tests also fail to evaluate out-of-band program component handling. No tape recorder we're aware of generates the out-of-band spectrum of a fast MC cartridge. Mind you, it isn't just tracing anomalies that create that spectrum. The generation of FM sidebands is inherent in the cutting/playback process itself.

All of these criteria are, of course, largely academic when the alignment for optimum lateral and vertical tracking geometry is neglected. It just isn't possible to listen critically to a preamplifier with the tone arm and cartridge incorrectly aligned. Much as we like to compare our results with those of other practitioners (including our subscribers), we really feel imposed upon when challenged to justify our findings to nonaligners with contrary opinions (e.g., Mark Davis). We don't believe the issue is negotiable.

Thanks to our expanded facilities, we were able to place all the preamplifiers reviewed below on open shelves at the same time and switch back and forth among them at will. All channels were adjusted for exactly the same gain at 1 kHz, from the cartridge terminals to the power amp input, within no more than 0.25 dB. At first we used an extremely costly "straight-wire" switching system for our listening comparisons and wasted an unconscionable amount of time. The switcher turned out to have subtle (but eventually measurable) shortcomings that put a barely perceptible haze over the sound and tended to homogenize the differences among the preamps. We then changed to A-B-ing by a lightning-fast manual plugging-and-unplugging technique that necessity forced us to master, and our conclusions are based on this cumbersome but far cleaner *modus operandi*. (Incidentally, we wonder about the switching system in the Mark Davis experiments. The design of an unquestionably straight-wire switcher is more than just a routine engineering problem.)

The reference speaker/amplifier system used in these listening tests was the Beveridge 2SW. With all its minor but frustrating flaws, it lays down a more lifelike sound field and resolves inner detail far more accurately than anything else known to us. There was really no other choice.

Do they really all sound different?

About half of the preamps tested followed the RIAA equalization curve so accurately that they tracked one another in frequency response within better than 0.2 dB. Phono input capacitance as a potential source of frequency response variations through cartridge inductance interaction was washed out by the 6-ohm MC cartridge used (even through the transformer). Volume levels, as we said, were accurately matched. So the basic requirements of the Mark Davis they-all-sound-alike school were satisfied in many, if not all, of our listening tests. And—we heard differences. In some cases small and subtle ones, in other cases laughably big ones. Therefore we now reaffirm our original conclusion that Mark Davis and the Boston Audio Society egalitarians are *wrong*. (Are-oh-en-gee, as one of our friends likes to emphasize it. But see also the lengthy correspondence on the subject in the *Box 392* column of our last issue.)

That said, we must admit that in certain cases two preamplifiers of different topologies can sound remarkably alike, even if not necessarily identical. We suspect, for example, that an open loop of limited bandwidth plus gobs of feedback to widen the response and reduce static distortions to triple-oh figures is a great "leveler" of sonic differences. (See also the power amplifier article in this issue.) Tracking these small differences in a blind A-B test requires extreme concentration that some auditioners find unusually fatiguing, so that they develop an I-give-up attitude even though they discriminated fairly accurately on their first few tries. Others manage to latch on to the sonic signature of either A or B or both and never make a mistake. We must emphasize that the difficult cases constitute the exception rather than the rule; anyone, for example, who can't immediately hear the difference between the latest Mark Levinson ML-1 and the comparably dead-flat GAS 'Thalia' should immediately consult a qualified otolaryngologist.

On to the reviews, then; and remember that as our test procedures become progressively more positive and revealing, there's less and less need for circumlocutory audio-word painting, so that the how-liquid-is-the-upper-midrange crowd may not be getting their money's worth here, even though they'll find out which preamp to buy and why.

Ace 3100

Ace Audio Co., 532 Fifth Street, East Northport, NY 11731. Model 3100 Stereo Preamplifier, $325 (with external power supply). Two-year warranty. Tested #P4016, on loan from manufacturer.

This rather Spartan but definitely audiophile-oriented unit uses the 4739 op amp chip in the phono stage. Since the incredibly successful Hegeman circuit module is built around the uA739, which is presumably the same thing (or almost the same thing), it's interesting to observe that the Ace 3100 also shares the Hegeman preamp's marvelously open, focused and detailed midrange. It's right up there in the near-SOTA category in that respect, but all is for nought on account of the highs. They're sizzly and nasty and well-nigh unlistenable. What a shame; this preamp *almost* makes it to the top at $325.

We don't think the RIAA equalization is the culprit, although its error curve is typical of the two-humps-and-a-saddle profile we discussed above. We'd call it ±0.8 dB, with one of the humps centering on 7 kHz or so, but that alone wouldn't make the highs sizzly. We measured 20 kHz harmonic distortion in the high single-oh region at two volts out, most of it coming from the line amp, not the phono stage. We don't think that explains the funny highs, either.

We'll just have to let Ace figure out what went wrong.

AGI Model 511A

Audio General, Inc., 1631 Easton Road, Willow Grove, PA 19090. Model 511A Stereo Preamplifier, $465. Three-year warranty; manufacturer pays all freight. Tested #7410167, on loan from manufacturer.

The "A" revision of this beautifully constructed and, from that point of view, very reasonably priced preamp differs in minor particulars from the original 511, so we retested it. Ours was the basic version having 33 dB of gain in the phono stage rather than Option H, which provides 40 dB of gain with the idea of sparing you the need for a transformer or head amp when using the higher-output moving-coil cartridges (not a low-noise solution, to say the least).

In our current reference system, the 511A exhibited a somewhat edgy, irritating quality that we don't recall in the 511, which we had considered quite smooth. Of course, the GAS 'Sleeping Beauty' Shibata cartridge through the Verion transformer will excite whatever transient problems exist in the input circuit. We heard a tiny burst of harshness every time there was a dynamic peak in the music. Also, there was an overall effect of thinness—or perhaps a lack of adequate fullness—in the sound, even though the RIAA equalization of the 511A is not far from perfect.

Don't misinterpret the above comments, though; this is still a very decent preamp, roughly of the same order of quality as the D B (although we tend to lean toward the latter in our preference). On the other hand, it isn't quite in a class with the Van Alstine, which in turn is still far from the winner's circle.

What's more, even if we happen to be wrong about these fine-tuned rankings, the Audionics (see below) is so clearly the top choice in the $400-ish range that the rest is academic.

Audionics BT-2

Audionics, Inc., Suite 160, 10950 SW 5th, Beaverton, OR 97005. BT-2 Preamplifier, $379 (with handles on front panel, $399). Three-year warranty. Tested #02350, on loan from manufacturer.

Here's the surprise and delight of the new preamp generation: a $379 unit that turned out to be one of the top three or four we had ever tested, at any price. Audiophiles with big ideas and limited budgets, rejoice!

The only preamps we know of that we clearly prefer to the Audionics BT-2 are the Hegeman HPR/CU ($645) and, one small step up from there, the latest Mark Levinson ML-1 ($1250). That's all. The latest Rappaport

PRE-1A ($755) has smoother highs and perhaps a wee bit better inner detail, but it's also thicker and more nasal in the midrange, where the Audionics excels. Others don't even come into the picture, meaning that the Audionics stands alone in its price range and then some.

Overall, we'd characterize the sound of the BT-2 as very open, smooth, beautifully defined, and extremely listenable. It lacks only the startling immediacy and super detail of our top choices.

In the laboratory, we found nothing important to criticize except the way the "new" RIAA playback characteristic is handled. From 1 kHz on up, the error is ± 0.0 dB all the way up to 30 kHz (wow!), but going downward it rises from -0.3 dB at 200 Hz to +0.4 dB at 39 Hz and then dips to -1.3 dB at 20 Hz. This is a *ripple* instead of a first-order roll-off, corresponding to neither the new nor the old standard. Strangest of all, there's a "filter defeat" switch (exactly according to our recommendation as discussed above) that increases the amplitude of the ripple by several dB instead of straightening out the response! So, all you bass purists, keep your cotton-pickin' hands off that defeat switch and leave it in "normal." (P.S. With the filter in, we heard no bass anomalies.)

In any event, we won't let this small blunder dampen our enthusiasm for this outstanding "best buy". Well done, Audionics.

D B Systems DB-1/DB-2
(follow-up)

D B Systems, PO Box 187, Jaffrey Center, NH 03454. DB-1 Precision Preamp, $397, with DB-2 Power Supply, $78. Five-year warranty; manufacturer pays return freight. Tested #1271128/2271128, on loan from manufacturer.

Since the D B still keeps cropping up in some SOTA discussions, we figured we might as well sock it into our current reference set-up and give it another listen.

Well, it still sounds a little hard and zippy. It doesn't actually have a nasty edge; it's not really unpleasant or irritating; it just isn't in the super category. We'd rate it maybe one small notch below the Van Alstine and on a par with, or very slightly ahead of, the AGI 511A.

One very interesting thing about the DB-1 is the way it interfaces with the D B Systems DB-6 power amp. The latter puts a very peculiar step into the leading edge of a square wave passing through it, sort of like a smooth bite taken out of the leading corner. When the square wave is first passed through the high-level stage of the DB-1, it's slowed down to the point where it no longer excites this behavior in the DB-6. Other preamps we've tried have no comparable "corrective" effect. We suspect there are some elusive feedback-related transient problems going on in D B equipment that may account for what we hear.

Incidentally, driving the DB-6 with the DB-1 didn't endow the former with more transparency, definition and spaciousness. So we're back to square one with D B—we find their stuff good but not great and we still don't know why.

GAS 'Thalia'

The Great American Sound Co., Inc., 20940 Lassen Street, Chatsworth, CA 91311. Model A-801 'Thalia' Servo-Loop Preamplifier, $299. Five-year warranty; customer pays all freight. Tested #A-801439, on loan from manufacturer.

We don't like this one at all, although its behavior on the laboratory bench is exemplary. Close to perfect RIAA equalization, beautiful reproduction of RIAA-preemphasized square waves (very fast rise time!), harmonic distortion of the order of 0.01% at 2 volts out throughout the audio range (even at 20 kHz!), you name it—everything is hunky-dory.

Except the sound. The high frequencies are badly smeared. The midrange is thick and poorly defined. And there's a pervasive haze or veil over the whole sonic presentation, making you want to wipe it off or tear it away. We're not saying that any of this is irritating or fatiguing. Just inaccurate. The design, we're told, is strictly P.B. (Post-Bongiorno), so we can't even talk about the taste for zabaglione slathered over the sound . . .

If nothing else, this proves the time has come for our new measurement procedures. It's frustrating to see such a chasm between the lab and the listening room.

Hafler DH-101

The David Hafler Company, 5817 Roosevelt Avenue, Pennsauken, NJ 08109. Model DH-101 Stereo Preamplifier, $299.95 wired. (In kit form, $199.95.) One-year warranty; manufacturer pays return freight. Tested #1805801, on loan from manufacturer.

Here it is, audio fans—the return of The Wizard of Cheap-but-Good. Dave Hafler, the man who conceived the original (tube) Dynakits, has come out of semiretirement and is bent on showing the world that the best sound money can buy doesn't necessarily mean a whole lot of money, not even in the solid-state era. He enlisted the very knowledgeable Ed Gately to design the circuit of a totally new and different preamp, complete with tone controls (no stripped-down little black box for these classy gents), and they're practically giving it away.

Well, is it SOTA or is it a piece of junk? The answer to that is a resounding "Yes!"

When we first listened to the DH-101 through our reference system, we were amazed. Other than the Mark Levinson and the Hegeman, no other preamp we had ever tried was even in the same league. Completely open, spacious, uncolored, focused and detailed, the sound was a joy and, at $300, an economic miracle. After long and agonizing A-B-ing, we decided that the Hegeman had definitely more authoritative bass and perhaps a shade more definition and immediacy, and that the Mark Levinson was the best of them all—but that was it. Everything else sounded less good than the Hafler, including the excellent Audionics. We then left the DH-101 "cooking" on our equipment rack with the power on (we do this routinely when testing low-wattage equipment) and turned our attention to other things.

Coming back to it a few weeks later for a recheck, we found the DH-101 sounding totally different. Gone was the live, transparent quality; it seemed that some kind of lid had been clamped on the sound. Not that we heard anything really bad, let alone obviously defective, but the super performance that had excited us was unquestionably gone, leaving the preamp one small notch above the GAS 'Thalia' in overall sonic accuracy. We then turned it off for a few days to let it "rest" and came back to it again. This time it sounded a little better but nowhere near its originally established par, and right in the middle of playing it went blah again, actually sounding a little *hard* (a la D B) for the first time. That did it.

Obviously what's happening in the Hafler preamp is that certain critical components change their correct value or response as they are burned in. In other words, El Cheapo strikes again. There are a number of electrolytic capacitors in the signal path; they may well be the main culprits. We aren't sure. We do like the circuit itself; it's quite original in concept and makes a lot of sense. But maybe it can't be put together as inexpensively as Dave and Ed originally figured. Too damn bad. (We'll try to take a look at a second sample to see if the same thing happens. One other user we know had a similar experience with the DH-101.)

Our bench tests, which took place before the decline and fall, revealed nothing we could even mildly complain about, except possibly the RIAA equalization. The Hafler is one of the two preamps covered here that follow that "new" characteristic (the other is the Audionics); no defeat switch is provided for the built-in low-frequency roll-off, but the bass control can be used in a pinch as a substitute. The trouble is that the topology seems to fall into the aforementioned hump-and-saddle trap, at least on the low end, with a very broad dip of 0.5 to 0.6 dB centering on 150 Hz—a bad place. On the other hand, the equalization is almost perfect from 1 kHz on up. And everything else —THD, square waves, all the routine stuff—is absolutely shipshape. One of the DH-101's claims to unique exellence is positive/negative pulse symmetry; the claim is true but the test is passed by many other units.

What *is* unique about the Hafler, at least among the preamps we've tested, is that it can give you an electric shock. Nothing lethal; just a nice, juicy tingle. Two large bypass capacitors on the AC line (0.01 microfarad each) are responsible for the leakage current that does this. Their purpose is to filter out RFI, but they do make the chassis rather exciting to touch if you happen to be earthed at the same time.

Of course, everybody knows that it's difficult to shock people nowadays; perhaps even a preamp that sounds different on Tuesday than on Monday won't do it, but a milliamp or two of leakage current definitely will.

Hegeman HPR/CU

Hegeman Audio Products Inc. (Hapi), 176 Linden Avenue, Glen Ridge, NJ 07028. Model HPR preamplifier with Model HCU control unit (incorporating power supply for HPR), $645 complete. Two-year warranty. Tested #103/101, on loan from manufacturer.

"I'm a lazy engineer," confesses Stew Hegeman, who at 65 has three times as many original audio designs to his name as any two hyperactive 32½-year-old engineers in the business. His laziness, he explains, caused him to design just one low level amplification stage, to be used repetitively in all of his current products. It's built around the uA739 op amp chip, compensated just so, and features tremendous bandwidth and high linearity, without the need for excessive feedback. (The uA739 is a true class A amplifier and probably the only genuine analog chip around.)

Stew takes two of these circuit modules, puts a *passive* RIAA equalization network between them, and that's one channel of his HPR phono strip, which goes right next to or under your turntable. The HCU control unit, built around one more of these modules per channel, then goes anywhere you have room for it, as long as the HPR's umbilical cord reaches the HCU's built-in power supply. It's really a very simple arrangement, and there's good logic behind it. The HPR has near-zero input capacitance, and minimizing the cable length between the arm and the input helps maintain that low figure. The low output impedance of the HPR then makes the cable capacitance between it and the control unit irrelevant. We don't think this chassis division is the secret of the Hegeman preamp's excellence, but it happens to be a very nice idea.

The HPR/CU combination turned out to be the most accurate-sounding preamp we had ever tested with the exception of the very latest version of the Mark Levinson ML-1. The Hegeman reproduces a sound stage of realistic depth and uncanny immediacy at the same time, with a genuine see-through quality in the midrange and highly focused inner detail. It's just a clearer and better defined sound than is available through other preamps. The highs are a little bit intense (not hard or irritating, though —far from it), possibly because Hegeman deliberately inserts a "corner" at approximately 21 kHz to stop the high-frequency slope of the RIAA curve, whereas in the cutter-head preemphasis network the corresponding corner is more likely to be at 40 kHz or thereabouts. (See also our earlier discussion of this above.) The bass is magnificent; Hegeman believes it's because he carries the equalization down to practically DC. (This is a whole can of worms that we won't open here now.) Another interesting thing is that between about 80 Hz and 6 kHz, the RIAA equalization error is actually zero—a very rare phenomenon, especially in the midrange. Does that have anything to do with the freedom from colorations? We refuse to speculate.

The distortion figures of the Hegeman are far from spectacular. High single-oh and point-one-something are typical on most static measurements. Obviously it isn't a feedback-happy design and just as obviously it has low dynamic distortion. Amen.

We must add that Hegeman Audio Products is a very small and only recently formed electronics company, with limited production facilities. We witnessed the slow and painful debugging of preproduction units such as ours, and we haven't seen an absolutely final, this-is-it production model yet. In other words, our endorsement of this exciting new product is necessarily conditional.

But if someone asked us right now which is the best preamp in the world at a three-figure price, we couldn't give any other answer but the Hegeman.

Linn Moving Coil Preamp

Audiophile Systems, 5750 Rymark Court, Indianapolis, IN 46250. Linn/Naim Type PNAG moving coil preamp with Type NAPS power supply, $250. Tested #0023/240V, on loan from distributor.

This Scottish-made pre-preamplifier for MC cartridges has been hailed in some quarters as a Verion beater, an assertion that (a) implies a somewhat casual attitude toward the laws of physics and (b) is untrue. (See our follow-up review of the Verion transformer below.)

The Linn happens to be a very nice-sounding low-level amplification module, preferable in our opinion to the Mark Levinson JC-1 and the equivalent System D for the ML-1. We believe it introduces less veiling and coloration. On the other hand, the Verion transformer introduces no character of its own whatsoever and presents a much more natural and balanced sound than the Linn. On transient peaks, the Linn is a little hard and glassy by comparison, and in noise level there's simply no comparison.

Because no amplifier is as good as no amplifier at all.

Mark Levinson ML-1

(follow-up)

Mark Levinson Audio Systems, 55 Circular Avenue, Hamden, CT 06514. ML-1 Preamplifier, with plug-in System A3X, $1250. Five-year warranty; customer pays all freight. Tested #2221 (with factory updates), owned by The Audio Critic.

We believe we've found the root of the ML-1 controversy. Some audio purists assert that the ML-1 is close to perfection—and they're right. Others find the ML-1 either hard or closed-down in sound—and they're right. How come? It's that damn volume control.

To guarantee accurate tracking within a small fraction of a dB, and to exclude the possibility of various volume-control-induced noises and distortions, Mark Levinson uses a ridiculously expensive wirewound potentiometer made by Spectrol. The original design of the pot was for servo control, not audio. A number of these super pots turned out to be defective in various exotic ways. Some of them started out with barely perceptible flaws, if any, and then gradually went sour. We had one of these. We couldn't quite understand why we had started to have less and less respect for our ML-1. Maybe we're becoming more critical, we thought, and our reference system is getting better. Well, it was the Spectrol. By the time we finally took a close look, it was making square waves ring with a leading-edge spike that went right through the ceiling. And the highs were biting like sharks.

In furnishing us with a replacement pot, Mark Levinson skipped a step. We were given the very latest, corrected-to-the-nth-degree Spectrol that will go into production units only beginning in April. The last version before that, we're told, was also quite trouble-free though not absolutely guaranteed to be so. The early model (that was ours) is highly suspect. You can recognize it by its completely smooth housing, without any external resistors soldered on it. If you have one of these, get in touch with the company and they'll take it from there. In fact, if you have any reservation at all about the volume control (or any other part) of your ML-1, don't hesitate to speak up. The whole point of insanely expensive equipment like this is that the maker can't afford to let you down. The Mark Levinson people are ready to lose time and money to update your preamp if it doesn't meet specs. When in doubt, ask them. They've got all the serial numbers of the bad guys and they want them back.

At the same time as our new volume control was installed, a few other updates were effected. These included new line driver modules (the latest are the best), the new A3X export phono cards (set for a gain of 33 dB to go with our Verion transformer), a couple of new resistors, plus some changes in component location. All of these modifications are standard on current production units except the A3X cards; the domestic A3 cards differ by one resistor and a few dB more gain. The important thing is to have the latest System A, whether domestic or export, as the earlier ones had an RIAA error of +0.4 dB or so humping up in the 10 to 12 kHz region, whereas the new ones are extremely flat (we measured ± 0.15 dB from 35 Hz to 35 kHz and -0.7 dB at 20 Hz) and continue their high-frequency slope at least three octaves higher than the old ones. (Forget about System D. The Verion transformer blows it away.)

With these changes our ML-1 reached new and unsuspected sonic heights. We had never hoped to find a better preamp than the Hegeman, but this one is better. The ML-1 sounds somehow more controlled, tighter and cleaner, with even greater immediacy and sharper focus. By comparison (and only by comparison) the Hegeman "blooms" a little. What we said about the ML-2 power amplifier is equally applicable here: next to the ML-1 all preamps have "hair" on their sound. The definition obtainable with the ML-1 is almost scary; things like double-tonguing in trumpet passages can be heard with an etched clarity beyond all expectation. Of course, some people may prefer that trace of bloom in the Hegeman (we suspect it has to do with a bit of innocuous second harmonic distortion, of which the ML-1 has a whole order of magnitude less—*and* without feedback instabilities); if so, they're entitled to their opinion and they've just saved themselves $605.

That's the permanent trap Mark Levinson has chosen to build for himself: unless he delivers the best there is, period, his prices make no sense at all. As far as the latest production version of the ML-1 is concerned, we believe Mark has escaped once again.

Rappaport PRE-1A

(follow-up)

A.S. Rappaport Co., Inc., 530 Main Street, Armonk, NY 10504. Model PRE-1A Stereo Preamplifier (with PS-1 external power supply), $755. (Also available with internal power supply as Model PRE-1, $620; with internal power supply and without tone controls as Model PRE-2, $520.) Three-year warranty; manufacturer pays all freight. Tested #1481-01/#2007, owned by The Audio Critic.

Having been apprised of certain internal refinements in the Rappaport, we traded in our early (almost prehistoric) sample and inserted the latest production version into our comparison lash-up. It did extremely well, giving all comers a hard time in A-B tests, but this time it didn't end up among our final recommendations.

The only flaw of the PRE-1A is a barely perceptible thickness—or call it nasality—in the midrange and the uppermost region of the bass. It's difficult to zero in on but it's there all right; you can hear it on brasses, for example. The result is a slight recession of realism and immediacy, a kind of homogenization, at least in comparison with the Mark Levinson and the Hegeman. The Audionics, too, has just a bit more midrange openness and transparency, but the Rappaport beats it on the top end. In fact, the Rappaport has become ultrasmooth and listenable in the upper octaves; we prefer it even to the Hegeman in that respect (though not to the Mark Levinson). All in all, a marvelous preamp still, barely squeezed out by the new generation.

It must be pointed out that, among the super preamps, the Rappaport has the least accurate RIAA equalization curve; the error is +0.7 dB in the 5 to 10 kHz octave and the high-frequency slope is flattened out only an octave or so above the audio range. The funny thing is that this ought to result in a subjective characteristic tipped toward the top end—but, if anything, the sound of the PRE-1A can only be called bassy, although the 20 Hz response is -0.7 dB and the midbass is dead flat. So the little anomalies we heard have nothing to do with amplitude response (Mark Davis, please note). On static distortion measurements, the PRE-1A appears to be impeccable.

Currently we also have a sample of the new PRE-2 (same thing but with internal power supply and no tone controls), but it arrived too late for these tests. Our superficial impression of it is that it's ever-so-slightly better than the PRE-1A, possibly owing to the simpler, more directly wired signal path. We plan to report on it in the next issue.

One thing you can be sure of. This isn't Andy Rappaport's last preamplifier. At the ripe old age of 20, he speaks of the PRE-1 fondly as his good old preamp and is working on something so far out he doesn't even want us to talk about it.

Van Alstine Model One

Van Alstine Audio Systems, Inc., 2202 River Hills Drive, Burnsville, MN 55337. Model One direct-coupled stereo preamplifier, $600. Three-year warranty. Tested #173101, on loan from manufacturer.

This is a very decent preamplifier, although it doesn't quite live up to the mystique with which its makers try to imbue it. We have a private communication from Frank Van Alstine that speaks of "our design—no feedback, no TIM, no phase shift" but the circuit details are kept very hush-hush and the epoxy-encapsulated plug-in modules withhold all information except name, rank and serial number. We seem to recall some mutterings about hybrid IC's, especially made for the Model One. We wonder.

The sound of the Van Alstine is beautifully open and essentially neutral, but there's a certain degree of grainy hardness and zippiness that keeps it out of the winner's circle. At first we thought this was due to a defect, as we had noticed too much DC offset in one channel; that was immediately fixed by the manufacturer and the whole unit rechecked. Well, it still sounds just a tiny bit edgy, grainy and irritating. Not unlike the D B, although the Van Alstine sounds more solid and authoritative on the whole. We'd rank it somewhere in the middle of the group reviewed here and unequivocally behind the Mark Levinson, Hegeman, Audionics and Rappaport.

In the laboratory we admired the almost incredibly accurate RIAA equalization, found everything all right on static distortion measurements, and were baffled by a weird *trailing-edge* spike that the phono stage puts on RIAA-preemphasized square waves. Since circuit details aren't available, we can only speculate about the relationship between this transient-handling peculiarity and the slight edginess of the sound.

None of this should be taken as a put-down of the Van Alstine; if we had nothing else to compare it with, we could live with it contentedly. The nature of our business is such, however, that we do have something else to compare it with.

Verion Mark I

(follow-up)

Verion Audio Inc., 75 Haven Avenue, Mount Vernon, NY 10553. Stereo Pickup Transformer Mark I, $375. Five-year warranty. Tested #1S224598 and #1S329497, owned by The Audio Critic.

After a year of shuffling moving-coil cartridges in and out of our reference system, and experimenting with devices to match them to our phono input, we want to make it loud and clear where we stand on this subject in general and on the Verion transformer in particular.

"Of course, any transformer in the signal path is a no-no," writes one of our correspondents, thus epitomizing the untutured folklore of audio-store cowboys. Little does he suspect that a transformer is actually *the* theoretically ideal impedance-matching device and that all you need in the case of an MC cartridge is impedance matching, as the energy coming out of the cartridge terminals (voltage times current) is certainly enough to drive any phono input circuit. The fact that the audio industry has traditionally produced crummy transformers has nothing to do with this fact of physics. It so happens that making a textbook-perfect transformer is extremely costly.

First of all you have to make sure that the core can't possibly saturate at the highest signal levels anticipated. That usually means that the transformer must be made bigger than anyone figured in his wildest dreams. Then you must deal with nasty problems like leakage inductance, shunt capacitance and hum shielding. All that involves a fairly sophisticated technology but no defiance of the laws of nature. Once you have placed all reactive interactions with both source and load well outside the desired bandwidth of the transformer, and kept externally induced hum and noise out of the signal path, you have a perfect passive device that *must* let more information through within its bandpass than any electronic circuit, for the same reason that empty eyeglass frames *must* let more light through than eyeglasses. There's no point arguing about this around the water cooler; it isn't negotiable.

What Mitchell Cotter did when he designed the Verion Mark I was simply that he went all the way and ended up with a complete and sufficient transformer (for phono signal levels and a bandwidth of approximately 80 kHz) rather than an underdesigned and incomplete one as others did historically before him. As a result, the Verion simply wipes out pre-preamps and head amps with respect to noise level and handily outperforms them with respect to signal fidelity—not because Mitch is a magician but because the laws of nature are on his side.

We know of absolutely nothing the audio purist can put between his MC cartridge and his preamp input that will let the signal through with as little audible alteration, either additive or subtractive, as the Verion transformer. Not the Mark Levinson JC-1 or System D, not the Rappaport MC-1, not the Linn, not the DB-4, not the MAS 1, not the Dayton Wright DW 535, not the GAS 'Goliath'—nothing. Make sure, though, that you get the correct strapping for your particular MC cartridge (P, S, PP, or X, as specified in the Verion literature), otherwise you'll lose a couple of dB of signal-to-noise ratio and dynamic range.

And if you're buying an expensive preamp (whether reviewed by us or not), don't buy it with the moving-coil option if you have a choice. The extra circuit won't be as good as the Verion. Mother Nature says so.

Recommendations

When it comes to preamplifiers, our usual disclaimer about not having tested them all begins to sound a little hollow. We've tested an awful lot of them, and it would take something quite new and special to replace the choices below. Not that it can't happen; we just don't expect it to.

Best preamplifier so far, regardless of price: Mark Levinson ML-1 (latest updated version only—see review).

Close to the best at a much lower price (and, incidentally, the next best regardless of price): Hegeman HPR/CU.

Best preamplifier per dollar: Audionics BT-2 (for absolute ranking, see review).

Best way to play moving-coil cartridges: Verion Mark I.

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The World's Best Piano Record?

Beethoven: Piano Sonata No. 23 in F Minor, Op. 57 ("Appassionata"). Ikuyo Kamiya, pianist. Japanese RCA, 45 RPM Direct Cutting Mastering Lab Series RDC-4 (imported).

Editor's Note: **Even though our in-depth audiophile discography is scheduled only for the next issue (Number 6), we couldn't wait to tell you about this one.**

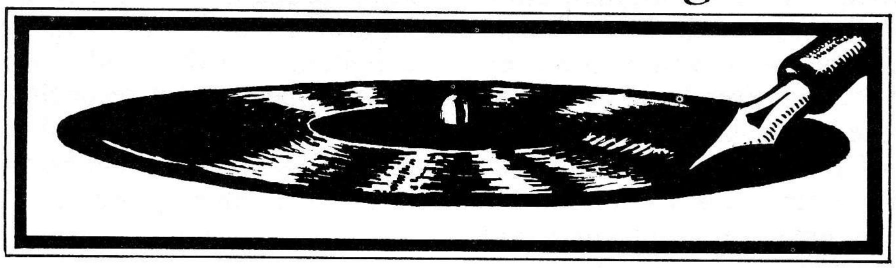
So you have big speakers, a zillion-watt amplifier and a sensational moving-coil cartridge, huh? How would you like to hear a Bösendorfer Imperial concert grand right between those speakers, big as life? It's more impressive, more solid, more authoritative than any Steinway and it's so real on this 45-RPM direct-to-disc record that it's scary.

We don't know of any other piano recording with this kind of dynamic range, presence and transient fidelity. For once, there's no shattering whatsoever on the upper octaves, even in fortissimo passages. (Of course, you must have the lateral and vertical tracking geometry of your cartridge trimmed in to the nth degree.) And, believe it or not, it's a multi-mike job! Which means that it doesn't quite possess the ultimate degree of spatial coherence, but with this kind of piano sound, who cares?

The performance by Ms. Kamiya is technically brilliant, even musically exciting, but lacking in the emotional subtlety and intellectual profundity that great Beethoven playing requires. A good ear, lots of energy, a terrific pair of hands, but a little on the cold side—that was the consensus of our staff. Somebody suggested that what Ms. Kamiya needs is a Hungarian boyfriend.

Okay, so it isn't one of those Schnabel albums. But it does sound a great deal better.

—Ed.



Editor's Note: Max Wilcox, our absentee Associate Editor, is back at last, with further evidence of his evolution from Establishment producer and organization man to free-lance independent and reborn audio purist. That little bulge you see under his jacket is yet another Grammy award, this time for reproducing Rubinstein's astonishing farewell album of Beethoven and Schumann piano works. (Eat your hearts out, all you direct-to-disc producers of The Saint Stanislas High School Band and other audiophile goodies.) The article that follows isn't exactly a sequel to his two previous ones, but we'll let him explain why.

Less Is More

By Max Wilcox

This article was originally to have been the third of the series on why records sound the way they do, and was to have taken a recording session from its conception to the final test pressing. A lot of it has been written and we should get to it in the next issue. In the meantime something much more pertinent to this publication has been happening in my studio work, and since there has been time to experiment, record, mix and evaluate the final results, I'd like to tell you what happened.

Last fall I embarked on two full weeks of recording for New World Records at Columbia's 30th Street studio. The week following the New World sessions I moved across town to RCA's Studio A for four sessions of Brahms piano music, played quite extraordinarily by Richard Goode for Desmar Records, and the remainder of the week was devoted to Tashi playing Mozart and Beethoven trios. I can say without exaggeration that, aside from the exceptional music-making that took place, these were probably the most technically provocative weeks I have ever spent as a record producer.

What could cause all this excitement after eighteen years of producing records? Well, it all had to do with microphones and microphone placement. And it had to do with—don't faint—*less* rather than more. In fact, the vast majority of the recordings made were recorded with only a pair of omnidirectional Schoeps/Studer "Colette" microphones. In the case of the Mozart and Beethoven trios recorded by Tashi, a pair of Cambridge C-5 ribbon microphones were added to the omni pair, for the cello and clarinet in the Beethoven and the viola and clarinet in the Mozart, to achieve an even greater degree of warmth and naturalness in the reproduction of those instruments.

* * *

The first session during which I tried the omni pair was for a record called *New Virtuoso Music, Vol. II* for New World Records (the American music recording company funded by the Rockefeller Foundation). The performers were Harvey Sollberger (the avant-garde composer and virtuoso flutist) and the vibraphonist Claire Hendricks. The work was a piece by Sollberger called "Sunflowers" and it was scored for alto flute, flute, piccolo and vibraphone. The various flutes, all played by Sollberger, were used in alternating sections of the music, so there was great range of color and dynamics from this duo of instrumentalists.

The microphones were placed about six feet apart at a height of seven feet and the players were about five feet away from the microphones. After listening to the players rehearse in the hall I walked into the studio to hear how it sounded over the microphone setup. Would the presences of the instruments match, would there be enough separation, would the omni pattern sound too diffuse? All of these questions ran through my mind. How did it sound? Terrific! Sollberger listened to a short test and said it sounded better than he thought he sounded in the hall. I had carefully placed the two performers about 3½ feet apart, which meant that the flute was essentially in front of the left microphone and the vibraphone was in front of the right microphone. Because of the omnidirectional pattern of the microphones both players had a full stereo pickup, yet the flute was clearly left of center and the vibraphone was solidly right of center. Any disparities in presence could be corrected by moving the players until the balance and presence were correct.

"Sunflowers" was completed to everyone's satisfaction and after lunch we returned to the studio to record a work by Robert Morris called "Motet on Do-dah". It is scored for flute, piano and double bass, and was played by Sollberger, Daniel Shulman, piano, and Donald Palma, double bass. Aha! Much more complicated balance problems. It was clearly a prime candidate for a fancy multimike setup. Well, I didn't weaken! We moved the players around (being careful to keep the flute in the same position and perspective as in the morning), removed the lid from the piano, and added one omni microphone about three feet from the double bass (panned on the console to correspond to his position in the group). The results were again wonderfully natural. The additional omni on the bass did not make the bass sound louder or more present than the others, but only served to correct the balance. A cardioid (directional) microphone on the bass would have brought him forward and spoiled the naturalness and depth. Again the players were delighted with the results, which is rather rare in chamber music recordings where the balances are so delicate and each player tends to feel his pickup could be just a little clearer, just a little more present.

* * *

The following week's New World sessions were performed by a New York early-music ensemble called the Federal Music Society, playing American music of the Federal period on antique instruments (not reproductions). There were woodwind septets, quintets and trios, in addition to works for violin and piano, piano solo, and voice and piano. It was quite a unique opportunity to try purist microphone techniques on a wide variety of instrumental and vocal combinations, and I was frankly fascinated to see how it was all going to work.

Needless to say, I had a small army of microphones close at hand in case this more-than-a-little suspicious technique didn't really work. Well, it worked. The lesson I had learned the previous week was as simple as: move the players until it sounds right and only add an extra microphone when a balance really cannot be corrected any other way.

First we recorded several pieces for 30-piece orchestra and chorus. The orchestra was picked up by an omni pair left and right in front of the orchestra, an omni pair on the woodwinds and an omni pair on the chorus of 12 singers who were placed behind the woodwinds. If the chorus had been larger, they would have needed less microphone help, but 12 singers cannot dominate a 30-piece orchestra without their own pickup.

The next day we recorded the various small chamber groups, and I never used more than two omni microphones, with an occasional single helper microphone on a double bass or snare drum.

The major undertaking was a recording of an opera, a delightful curio from the early 19th century called *The Ethiop* by Raynor Taylor. Now three vocal soloists were added to our orchestra and chorus. Here we used an omni pair for the general orchestra pickup and a 12-foot-high pair of Schoeps/Studer cardioids for the woodwinds. The helper microphones we had put up for the double bass and percussion section proved to be unnecessary and were not used. The chorus, which was several feet behind the orchestra on risers, was recorded with another omni pair of Schoeps/Studers. As I mentioned, choruses are notoriously difficult to balance with an orchestra, and from the conductor's podium in the studio the chorus sounded very far away. Using an omni pair about 11 feet high in front of the 12-voice chorus allowed us to establish a balance and yet keep a similar perspective on the overall sound of chorus and orchestra. The vocal principals were placed about 10 feet to the rear of the conductor and sang, facing the orchestra, into their own stereo pair of cardioid Schoeps/Studers. The microphones were about 3½ feet apart and the singers were about 5 feet away. The distance ensured a spacious vocal pickup and the directional pattern (with the dead side of the microphone facing the orchestra) gave us adequate control over orchestral leakage into the soloists' microphones. If time had permitted we would have experimented with placing the soloists beside the conductor, but we had three sessions for a great deal of music, and regardless of their physical position the vocalists needed their own microphones. Actually, in recording sessions, vocal soloists are almost never recorded in concert position (beside the conductor) because a separate pickup gives the engineer and producer much more control over the balance and eliminates the problem of orchestral leakage into the vocal microphones.

* * *

Well, the following week I got to deal with that problem too. After finishing the New York recordings, I flew to Chicago to produce the audio for Unitel/Munich as they videotaped three new programs with Sir Georg Solti and the Chicago Symphony. These programs are shown all over the world, and in most large cities the sound is simulcast in FM stereo over a local FM station.

During that week we taped an all-Strauss program (including *Death and Transfiguration*, *Till Eulenspiegel* and the *Four Last Songs* sung by Lucia Popp), a Russian program of Prokofiev's *Classical Symphony* and the Shostakovich *Symphony No. 1*, plus an hour-long program of the major orchestral pieces from Berlioz's *Romeo and Juliet*.

All right, now what was I going to do with microphones? Last year the Solti/Unitel programs were made with 14 to 16 Neumann U-87 microphones and Sir Georg was delighted with the sound. How could I use my experience of the previous weeks to improve upon the sound, without compromising the control necessary to match the sound to the ever-changing close-up pictures of various orchestral sections that flash across the screen? These pictures must be supported by sound that bears some relationship to the image seen by the eye, and yet one cannot break up the sound into a series of close-up recordings of the orchestra that have no overall perspective and homogeneity. My solution has been to balance the orchestra the same way I would in a regular recording session and, if all the instrumental perspectives are right, the picture can do whatever it likes without ever seeming to be unsupported by the sound. A screen solely occupied by a snare drum playing softly must be accompanied by a clear recording of that drum, yet the drum must still retain its proper place toward the back of the orchestra in the overall audio perspective. Hence last year's 14 to 16 microphones.

The question was what microphones from last year's setup could be eliminated without losing the control necessary to support the video images. Well, first there was the problem of the hall. Orchestra Hall Chicago today bears little resemblance to the hall of Reiner's famous recordings. It was "modernized" in 1967 by the acousticians who gave you New York's original Philharmonic Hall, and its glories are but fond memories. In other words it has little resonance or luster, and filled with 3,000 people (as it is at the special concerts performed for Unitel's cameras) it becomes a definite recording challenge.

While still retaining the microphone setup features that had pleased Solti the previous year, I began to eliminate microphones. In rehearsal we found the timpani sounded best without a separate microphone, and the double basses sounded best picked up by the overall microphone on their side of the stage. There had been four omni microphones out in the hall the previous year and these were replaced by one cardioid pair facing *into* the hall. So now we had a percussion microphone and a harp-and-piano microphone as the only spot microphones. The remainder of the sound (all recorded this season by Schoeps/Studer microphones) was picked up by an omni pair left and right at the edge of the stage and 11 feet high, a cardioid pair 10 feet above the woodwinds, and a cardioid pair over the second violins and violas. In the *Four Last Songs*, Lucia Popp was recorded by an omni pair about 5 feet away, and panned full left and right to preserve the stereo image of the orchestra being recorded on the main microphones. The famous Chicago brass certainly needs no separate microphone to immortalize its stentorian tones, so we had a total of 12 microphones. The percussion and harp/piano pickup were used only during special solo passages, so essentially the sound was recorded by 10 microphones. Still multimike to be sure, but far fewer than are commonly used these days in orchestral recordings.

* * *

Of special interest was the cardioid pair facing into the hall that replaced the previous season's four omnis. Mixed at full level with the main microphones, they gave more spaciousness to the sound than we dared hope. I first tried this hall microphone technique (which had been used for years by DGG *Tonmeister*) during a Philadelphia Orchestra/ Unitel television production in June, and it made the Academy of Music sound very spacious. Not lushly resonant, mind you, but certainly spacious and unconstricted. This technique captures the reflections off the auditorium walls as they are returning to the stage in their longest period of decay.

Since the acoustic personality of any hall is determined by a combination of direct sounds from the stage and reflections from the walls, microphone setups that do not capture hall reflections in true proportion cannot accurately reproduce the acoustics as actually experienced within the hall. Using distant hall microphones pointed toward the stage gives a full, resonant pickup, but it also creates a time blur because the direct sounds recorded by these hall microphones arrive 35 or 40 milliseconds *after* they have been recorded by the stage microphones. The cardioids turned away from the orchestra and into the hall record only the *reflections* as they are heard from the stage, and the arrival time of direct sounds is therefore not upset.

This technique, first tried in Philadelphia, was equally effective in the similar dry acoustics of a packed Orchestra Hall. Both the Ormandy and Solti programs will have been telecast and simulcast by the time you read this. The sound you heard was not compressed or equalized in any way during my mix made from multitrack tape to two-channel stereo, and those of you who heard these programs in good FM stereo simulcasts heard a dynamic range not usually expected from a made-for-television production.

* * *

Since both Solti and Ormandy are under long-term contract to Unitel, I will have a yearly chance to make what I hope will be increasingly natural and technically sophisticated recordings. Coincidental (crossed pairs) microphone techniques were first written about by Alan Blumlein in the early 1930's, long before stereo became a commercial reality. Since his theories are based on phase coherence, which in turn is based on fundamental laws of acoustics, it is highly appropriate that many people are reinvestigating his techniques. Many of the Sheffield Lab recordings are based on coincidental microphoning, and my next project will be my own series of experiments with this technique. Schoeps is now preparing to market a special flat-frequency-response capsule (similar to the omnis and cardioids referred to here) in the figure-eight configuration on which Blumlein based his theories. Since I prefer omni microphones to cardioids, I have been hesitant to get my feet wet in coincidental microphoning. Omnidirectional microphones produce wonderful *mono* when used in coincidence, since they do not possess the directionality upon which coincidental microphoning is based. Figure eight gives you a cardioid-like pickup both to the front and rear, so the hall pickup of an omni is approached while still retaining directionality in front.

Of course my much-favored Cambridge ribbon microphones are figure eight, so perhaps what I really needed was a little mental stimulation to move briskly from the convenient to the possible! This summer I'll try some coincidental microphoning on the annual Ormandy-Philadelphia/Unitel production, and I'll report the results on these pages.

Since this article actually started to be written aeons ago as a history of a Tashi recording session, you can see how far things went astray. I promise in the next issue to tell what happened there, and how we went through the various steps that will result in an April release on RCA Red Seal.

Editor's Note: Is it a coincidence, is it our imagination, or could it possibly be the cumulative effect of this column? (No, it couldn't be . . .) The fact remains that, since the last go-around, we've seen a drop in the outrageous technical claims and denials of the laws of nature in national hi-fi advertising. In fact, the item below is the only one that has seriously stirred up our critical juices lately.

Marantz 940

Back in the 1960's, the name Marantz was synonymous with the world's finest audio equipment. Even today, the Model 7C preamp, the Model 9 power amp, the Model 10B tuner of that period sell at a substantial premium on the secondhand market. People like Mitch Cotter, Sid Smith, Dick Sequerra, Julius Futterman and others of their perfectionist bent were involved in the design of the equipment, and Saul Marantz saw to it that nothing left the factory with his name on it that wasn't the very best. The comedown after Saul was bought out by Superscope was almost immediate, but it took quite a few years before Marantz became indistinguishable in product quality, image and advertising from Pioneer, Sansui and other large Japanese hi-fi companies. Today the process is complete and Marantz's advertising is, if anything, more heavy-handed and "buckeye" than any other Japanese mass merchant's.

Their latest series of four-color double-page spreads ("Ultimately It's Marantz. Go For It.") is particularly shrill and obnoxious, and among these the one on the Marantz 940 speaker system (top model of the Marantz Design Series) really has us going up the wall.

The first paragraph of the copy announces "the sharpest, cleanest instrument definition you've ever heard from any speaker system." (Guess we'll have to trade in our Beveridge.) The third paragraph explains that the crossover network of the Marantz 940, by virtue of its six (count them) inductors and three level controls, is the most sophisticated *ever* and assures you of precise crossover points, smooth transitions and flat frequency response. And here we were, believing that the more complex the crossover, the more difficult it is to control those very characteristics; but live and learn. (Maybe they're compensating their network with all-pass filters, Papoulis sections or something, right? Fat chance, for $400 . . .)

But it's the sales pitch on the woofer, further below, that takes the booby prize. "The big bass drum is heard in all its glory because Marantz builds woofers with a rigid new cone material—rigid enough to withstand ten times the force that can destroy a light airplane. This superior structural strength enables the cone to move in an ideal piston-like motion, instead of bending. Which means a tight low-frequency response and uncolored sound quality." This is pure taurine excrement.

We must admit that it upsets us so much because we've heard it before; rigidity as an argument for woofer quality is one of the more common techno-illiteracies of today's audio scene. The truth of the matter is that rigidity is a *high-frequency* requirement, not a low-frequency one. The best woofer cones are soft, pulpy and lossy. "Piston-like motion" at, say, 100 Hz can be obtained with a woofer made of mucus; it has nothing to do with "structural strength" but with the ratio of the cone radius to the wavelength. It's when the woofer cone has extended *high-frequency* response (thanks to rigidity!) that you run into the problem of sonic colorations and need a complex, steep-cutoff crossover section to counteract the problem.

The ad also pushes the Marantz "Vari-Q" feature, about which we always had the gravest misgivings (it isn't new in this model). It consists of a removable foam plug, so you can convert a sealed-box woofer into a vented box. The trouble is that it isn't mathematically possible to convert a properly aligned sealed box to a properly aligned vented box simply by opening up a hole in it, without any changes in the driver design or the box dimensions. Either one or the other alignment will be wrong; in the case of the Marantz we'd expect both to be wrong. Removing the plug undoubtedly changes the speaker into an uncontrolled, high-Q boom box; in fact the ad talks about "the gutsiest low end for today's electronic rock."

From our consumerist point of view, the worst kind of hi-fi ad is the long-copy, pop-tech kind that has the surface appearance of educating the reader but is actually designed to mess up his head so he'll think that (a) he now has some solid technical information and (b) the product meets the technical criteria he has just learned about. This is that kind of ad. Yechh.

We just hope that every man, woman and child with the slightest interest in audio knows by now that Marantz is no longer *the* Marantz.

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