The COLONY AIR BEARING TURNTABLE
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Contents

AS WE SEE IT
The Truth Should Out..........................3

CHICAGO CES.................................5

EQUIPMENT REPORTS
The Shure V-15 V Cartridge...................10
The ABX Comparator...........................11
The Acoustat TNT Amplifier..................14
The B&O Beosystem 8000......................17

RECORD REVIEWS..............................20

MISCELLANY..................................22

THE LITERATURE..............................28

LETTERS.....................................30

AUDIO PUZZLER..............................37

AUDIO MART.................................39
The Truth Should Out

This issue contains a report on a truly ingenious little device called the ABX comparator, which takes the fraud out of subjective testing. It does this by making its own selection of source A or source B for each listening trial, without telling you which was selected. Only after all the tests will it reveal what you were listening to each time. "Score" sheets are provided so you can list your guesses, compare them with the cold, uncompromising truth, and file the results for posterity. Or better still, for the first hard evidence that has ever been presented that a lot of people can hear differences that cannot as yet be measured.

A couple of issues back, I said in this department that it was past time that we perfectionists either put up or shut up. We have claimed for so long that we could hear things the scientific establishment doubts we can hear, and disagreed so much about what we hear, that so-called subjective testing has lost the last vestiges of its credibility. This has not been helped at all by recent assertions that reproduced sound is fouled by the proximity to equipment or cables of metal, wood, concrete, carpeting and people, nor by claims that interconnecting wires must be dimensioned to within a ridiculous fraction of an inch to avoid total destruction of fidelity. We have even heard a report recently of reproduced sound being "dramatically" improved by the placing of a small container of water beneath the resting position of the stylus, EVEN THOUGH THE STYLUS NEVER CAME NEAR IT! This sort of thing raises audio above the level of science into the cloud-nine realm of mysticism, magic and voodoo.

I am not claiming that any of this is impossible. "Impossible" is a very strong word, often freely used these days by people with an overweening desire to be proven wrong. What I AM saying is that allegations like these are so hard for a scientific mind to swallow that anyone who makes them without at least offering some sort of proof that they ARE audible lays him- (or her-) self open to ridi-

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cule, derision, and out-of-hand dismissal as a full-blown crack-pot.

One of the things which makes these claims so hard for a scientifically-trained observer to swallow is the fact that the phenomena do not conform to the inverse-square law which controls all known force phenomena above the subatomic level. The I-S law, to refresh your memory (?!), observes that forces or energies which are not specifically directed (i.e., those which act uniformly in all directions) are reduced to one-quarter strength each time the distance from the source is doubled. The law reads like this: Intensity is inversely proportional to the square of the distance. The "law" works for light from a candle, heat from the sun, and sound from an omnidirectional (nondirectional) radiating source. But the proximity phenomena reported by some audiophiles seem consistently to violate the I-S law -- another reason why the technically-schooled view them with extreme skepticism.

Yet there are reported phenomena outside of audio which do not seem to conform neatly to the I-S law either. Mental telepathy seems able to act over vast distances without loss of carrying power, but then telepathy has never been scientifically proven either. Like the mystical tenets of organized religion, it has been proven to the satisfaction of believers but not to the satisfaction of nonbelievers. Few scientists believe in a white-robed, bearded God who sits in heaven to love, judge and answer prayers, but their stance is rarely one of disbelief, for it is even harder to prove nonexistence than existence. Most are doubters -- agnostics -- who are not convinced of the truth of either stance. That just about describes where I stand on many of the things certain audiophiles claim to be hearing.

There is no way that the ABX comparator can be used to prove or disprove the existence of a caring God. But it DOES offer hope of proving, for the first time, whether some of the less-credible observations of perfectionists have any basis in reality or are just magnificently-conceived self-deceptions. That is why, even though I feel that the device itself needs further implementation, as the jargoncrats would put it, I also feel it is the most significant contribution to audio that has come upon us in years.

But it has to be used. Few audiophiles would pay $500 for a gadget that might do no more for them than prove they have been deluding themselves all these years. But audio clubs could afford them, and should use them. So can and should every component manufacturer that REALLY wants to find out if its latest product is truly better than the competition. The device COULD cause more embarrassment in this world than the invention of the rattlefart, but whenever truth and conviction are at odds with one another, embarrassment for some is inevitable, good, and necessary for the advance of knowledge. The losers will be the dissemblers, the frauds, and those most skilled in the art of the autohype. The winners, ultimately, will be music and the rest of us who are interested only in the maximal fidelity of reproduced music.

So, I wish ABX well, I hope they prosper and -- mainly -- I hope the audio community takes full advantage of this, our first opportunity to prove that at least some of what we've been bitching and moaning about all these years does in fact exist and can, if pursued seriously, improve the state of the audio art. JGH
CES Stop Press

Our full report on the Summer 1982 CES high-end audio exhibits will appear in the next issue. But there are some items so timely that they must take priority over what was previously scheduled for this issue. One is our "Digital Dirties" item which appears elsewhere in this issue. The others follow here:

English tone-arm manufacturer SME, which made the arms Shure Bros. sold for use with their own and for other low-mass cartridges in the US, has gone under. There is talk of a bail-out by the English government, but it is no more than talk as of this time.

Mobile Fidelity records, a longtime skeptic of digital audio, is planning to release pre-recorded digital cassettes and discs of their Original Master recordings, as well as what they call videophile releases -- described as video art with high fidelity sound. The PCM tapes will be in the Beta format and possibly VHS also, and the discs will be in the Philips laser disc format.

With the retirement of "Stereo Review" Editor William Anderson, Audio Editor Larry Klein expects a freer editorial hand and is already planning a new section in the magazine devoted to perfectionist interests. Tentative title: "Tweaks". Scheduled for Fall publication is an article about the "esoteric audio" field, which will mark the first time SR has admitted that there are a lot of "underground" audio magazines out there.

Paul Messenger has been named publisher of England's leading audiophile magazine, "Hi-Fi News and Record Review". Personal friends report that he has mixed feelings about the new position, as his managerial role will remove him from contact with the components he has loved working with in the past.

Ivan Berger, of Berger-Braithwaite Labe, has been named Audio Editor of "Audio" magazine.

Music and sound, formerly of Willow Grove, Philadelphia, then of Los Angeles, CA, now of Huntingdon Valley, PA, is importing a $400 English turntable that looks like competition for the Linn. We have one for review.

Exhibiting in the US for the first time was a small English electronics company named "Valerie".

CES HIGHLIGHTS

In my view, the '82 Chicago CES was unusual by virtue of the generally high quality of sound, oftentimes at reasonable prices, rather than for any astonishing breakthroughs. Though there was the usual garish, unlistenable sound in a number of rooms, there were also a surprisingly large number of exhibits where the quality of sound was markedly better than I'm used to at home. There are, I think, three general developments responsible for this:

1) A proliferation of high-quality electronics. Names such as Krell, Bipolar Electronics Systems, BEL, Futterman, Electrocompaniet, Berling, and VPS all could be found in rooms with good sound. In particular, I found myself in rooms noticing first the quality of the sound and then the fact that they were using Audio Research electronics. It should be pointed out that none of these amplifiers are inexpensive; but the fact that good sounding electronics are converging towards a similar sound (open, clean, wide dynamic range, relatively uncolored) is certainly a good sign. Hurrah for the makers of good electronics! It's long
been my contention that even lesser speakers will sound quite good with excellent electronics (while perhaps not revealing all there is to hear) — conversely even the best speakers do poorly when powered by deficient circuits.

2) Widespread use of excellent turntables, along with the introduction of some new, high-quality ones. The new ones are the Pink Triangle, Colony Air Bearing (also seen in Las Vegas), C.J. Walker (an inexpensive model from England now in-house for testing), SOTA Sapphire, and the new-to-my-eyes Goldmund. Even if the Goldmund turned with lurching, 15-degree increments I would want to own one it's so beautiful; as it is, the Goldmund and it's tangential-tracking, servo-operated arm are reported to be one of the best information retrieval devices available. Virtually all of the new turntables are emphasizing isolation of the system from mechanical and airborne feedback; accurate rotation and low friction through close machining tolerances and fancy bearings; and effective dissipation of vinyl resonance through intimate contact between the record and something. In this last area there's every approach imaginable: from the Linn felt mat which must do pretty well, to the naked Goldmund "methacrylit" plexiglas platter, to the Oracle rubber mat that has the record screw-clamped to it; the list could go on and on. We're eager to receive these products in order to conduct some tests which will convince at least us of the validity or lack thereof of these different theories. It's ironic that only when non-mechanical information retrieval systems are finally upon us does the audio world really dig in and go all out for the information on our princi-

3) Numerous good loudspeakers. Acoustats, Roger West's Sound Labs, Fuselier, Vandersteen, GNP Lead Cylinder, Spica's SAW-1, Dahlquist DQ-10 (sounding maybe as good as I've ever heard with Audio Research electronics), and most ubiquitous of all, Quad's ESL-63's (which we can't seem to get to test, dammit) all put in good performances. I'm sure with the right electronics I could happily live with any of the above speakers — who knows, I might get so into the music I'd stop writing about audio! And that's just a partial list — there were many rooms I never even got to.

Interestingly enough, there was only one speaker disarming enough to invite the "willing suspension of disbelief" that we all seek out in sound reproduction as in the theatre: the Dahlquist-distributed MAGNAT speaker utilizing a plasma discharge driver. MAGNAT, a West German firm, makes a whole line of speakers using their own drivers of somewhat unusual design. A brief audition to these other speakers revealed nothing that in the hurly-burly of CES would warrant a second audition. But the Plasma/Magnats stopped me cold. Why? Nothing. There was nothing there: no speaker, no awareness of a record playing, just the music in the far end of the room being enjoyable for its own sake. Unlike the Plasmatronics Hill Type I (one of my favorite speakers), one was not specifically aware of hearing the best high end ever reproduced — one simply listened to the music. Speaking of the Hill Type I, this is the first CES Alan Hill has missed for maybe 4 years. We hope that his involvement in the fabulously inventive laser development work that Plasmatronics does to make money hasn't absorbed all his time and energy. (Speaking to him later in Albuquerque I learned that
Alan simply did not have the time to make this show but is planning to exhibit his speakers in Germany and at Winter CES in Las Vegas.)

The Magnat plasma driver, developed by Dr. Klein (also from Germany), operates on the same principle as the Plasmatronics: the heat produced by a plasma discharge forces rarefaction and compression waves in the air by way of thermal expansion; modulation of the plasma discharge, and hence of the rate of heat production, creates waves of heated air moving out at frequencies within and above the audio band forcefully enough to be perceived as sound. Since the "driver" of the air is heated air itself, the mass of the "driver" is reduced to little more than that of the molecules of air in the heated region. This heated air displays none of the resonances inherent in even electrostatic diaphragms, much less dynamic drivers, though there are problems that must be solved in plasma drivers in connection with the behavior and escape of the heated air (Above certain frequencies and sound pressure levels, the speaker becomes non-linear due to the inability of heated air to leave the immediate region of the driver in correspondence to the input signal).

The Magnat plasma discharge, however, is produced in an entirely different way than that in the Plasmatronics Type 1. Rather than an ionizing arc between highly positive and negative electrodes arranged in a roughly conical shape and bathed in a constant flow of helium gas, the Magnat possesses a needle-shaped center electrode totally enclosed by a spherical wire mesh ground grid. A 27 megahertz carrier signal is introduced to the center electrode at a voltage sufficient to create a high-frequency ionic discharge between that electrode and the wire grid. This is the plasma, and the superposition of the audio signal on the 27 MHz carrier frequency produces the thermal and audible effects referred to above.

This plasma-producing device is in turn surrounded by an acoustically transparent protective metal grid. The whole thing is about as big as a softball and sits on a stem like something out of Star Wars, all atop a roughly conventional cabinet which contains the lower-frequency drivers. Further comments on the system will await a review sample (which I think unlikely to be soon forthcoming) but I discovered an interesting thing further on at the show. Dave Wilson of Wilson Audio Specialties uses the low-end driver from the Magnat/Plasma system, a huge 18" affair with dual voice coils, for his $32,000 WAMM system (see issue V-4), in spite of its $500/driver cost — a fantastic sun, outdoing even the legendary Hartley 24" driver.*

There are a few other products which I feel need mentioning, considering the time I spent visiting with their proponents. Bruce Trigpen went through a very interesting, and dispassionate, explanation of the workings of the Colony Air Bearing Turntable, which he and Lew Eckhart developed together, working for the the Wayne H. Colony Co., a munitions supplier to the US government! Don't let that put you off — a turntable by any other name is still not a gatling gun. This turntable has gone about solving it's isolation and friction-related problems in a novel way: it uses air wherever it can to insulate vibrating and frictional components from one another.

* Skeptical readers comparing that $500 to the $32,000 and raising an eyebrow or two would probably be interested in an article on speaker economics. In most speakers, the driver cost by itself is minor. One example of a super high quality 3-way system with which I'm familiar has approximately $35 worth of drivers, $80 worth of capacitors and inductors in the crossover, and so much in the cabinetry that the speaker system can't be sold at its intended list price of $1100 and make any money!
The turntable platter, which is belt driven, sits on a stationary platform full of tiny holes. An air pump is connected to those holes and the machining of the platform and the platter is so even that the air can escape only by pushing the platter uniformly up about .007". That's plenty to yield a frictionless surface, even with a heavy mat on the platter (the spindle is exceptionally high to accommodate just such a mat). Audio Technica has come out with an accessory vacuum disc-clamping device weighing about 3.5 lbs - Bruce thought this would be right at the limit of what the air would lift (maybe too much). So the air provides the vertical, virtually friction-free bearing. Since the table is belt-driven, and the belt pulls the platter to one side, there must be a lateral bearing as well. This bearing is the spindle, which the platter slips over snugly. There is a bushing in the platter which needs occasionally to be relubricated with a very small amount of transistor heat sink silicone compound. Since the force of gravity is not active laterally, the only force against this thrust bearing is that provided by the motor. Compared to some other 'tables, this bearing might be considered inadequate (one, the ISOS, uses a bearing a full 3" long). I was assured by Bruce that the bearing was quite adequate; only testing will reveal any sonic effect from a low amount of bearing surface.

The really interesting thing about the Colony product, however, is the arm; it is a tangential tracking design which solves the problem of arm movement (handled in other tangential arms, the Rabco, the Goldmund, the Yamaha, etc., by motors and drives) in the same way a pivoted arm does: it uses the stylus following the spiral record groove. In a theoretically ideal transduction setup, the cartridge body, fastened to a tone arm of infinite mass to eliminate resonance and so that all movement of the stylus was translated accurately into an electrical signal, would hover weightlessly over the record groove suspending the stylus in the center of the groove so that it was free to respond equally to modulations on both sides of the groove. Since here on earth mass equals weight, the real cartridge and tone arm setup is very different from this ideal. In a conventional pivoted arm, the leverage provided by the length of the arm reduces the amount of flexing force applied to the cantilever in moving the arm -- across the disc and in response to warps. In straight-line-tracking arms, leverage is something to be avoided, as it tends to twist the "trolley" that carries the arm along its track, increasing motional friction. Even with the shortest arm, however, rolling friction in the usual trolley-design straight-line arm generally puts more flexing force on the cantilever than it sees from a pivoted tone arm. Moreover, that flexing force is exactly contradictory to our model above where the stylus is expected to sit effortlessly, and without bias, in the groove. Until recently, the most successful efforts to overcome this friction employed a motor to drive the trolley, and a switch to activate the motor whenever the arm's tangency trolley exceeded some predetermined amount. Here again, though, we're producing exactly what we don't want (tangency error) in order to accomplish what we do want: move the tone arm along the record. In a well-designed air-bearing arm, sliding friction is virtually eliminated so the cantilever need overcome only the combined inertia of the arm and cartridge. In the Colony, the tone arm is attached to a hollow tube which is machined to have a 5/10,000ths of an inch clearance from the inner tube it slides along. This inner tube also has little holes in it connected, to the same air pump, so again the air can escape only by holding the outer tube clear of the inner one. The result is a truly frictionless "track" for the
cartridge. You can bounce the tone arm (sans cartridge naturally) back and forth between the fingers of each hand 6" apart with almost no force. So the tone arm (with almost perfect tangency - I calculate a maximum of one minute of one degree potential error) literally floats the cartridge over the record. With no friction, the cantilever need overcome only the inertia (mass) of the arm in order to move it. In order to reduce mass, Colony went all out and applied the tracking force "counter-weight" by magnetic repulsion. A metal strip extending from the arm tube has a small, potent magnet at its end. This hovers out over a fixed bar magnet of the same polarity which can be lowered and raised, to change the amount by which the magnets repel each other. The repulsive force acts to lift the arm away from the record surface. The elegance of design and extremely thorough attention to detail is enough to revivify one's pride in American industry.

The one problem not solved is inherent to the design: since there is literally no connection between the turntable and the arm, other than the cartridge, any sudden lateral motion transmitted to the turntable (such as bumping the frame) will cause a radical disassociation of platter and arm -- groove skipping or the like. I'm dying to have one on hand to see if all this ingenuity and superior execution results in a product with superior sonic attributes.

The third product I'd like to examine in some detail is the ISOS turntable referred to above. ISOS is a company founded by two escapees from the Dahlquist firm, John Fink, and Frank Rizzello, and a third partner, Jim Worthington, a mechanical engineer who also brings financial expertise to the firm. They are another of the firms, like Oracle, SOTA (also known for their head amp), and Colony, whose principle claim to fame in the audio world is an all-out assault on state of the art turntable design. A key element in the approach of ISOS is the development of a patented approach to the platter itself. The platter on which the record rests is isolated from the drive platter by "a rigid (no information loss) mechanical high-pass filter" consisting of three contact pins out at the circumference of the two platters. The turntable is belt-driven, incorporating a somewhat better design of speed change from 33.3 to 45 rpm than is standard. It still involves changing the belt from pulley to pulley except that in this case there are stepped pulleys on both ends - the platter end and the motor end - so that the belt tension does not change. It might be pointed out that this revolutionary technique has been in use for decades on such sophisticated devices as drill presses, but it's still good to see it adopted for turntables, though I'm not convinced that any problems that might be caused by a slight bit of additional tension on a turntable drive belt.

The lateral and vertical bearings have tiny clearances and very close machining tolerances, as we've come to expect from high end 'tables. In any case, it's not primarily the design of the ISOS which I found interesting, but the demonstration of its sonic superiority to a Linn-Sondek. Those readers familiar with Linn's own demonstrations will know how the test goes (as well as the results): two turntables with identical (the same brand) cartridges and tone arms are set up, and selections are played on first one and then the other. I heard the Linn-Sondek demonstration several years ago against the best Denon of the time, and the differences were clearly audible, in spite of my inclination to prefer the Denon since it so closely resembled in design and performance my own SP-10 Mk II. There was more dynamic range, better preservation of inner detail, and some tonal balance differences which (con't on page 26)
**Shure V-15-V Cartridge**

Moving-magnet stereo cartridge. Frequency range 10 to 28,000 Hz. Separation 25 dB or more at 1 kHz, 18 dB or more at 10 kHz. Output 3.2 mV at 1 kHz at 5 cm/sec. Stylus: Nude hyperelliptical, .2 by 1.5 mil. Tracking force: 1.0 gram at tip (0.5 grams additional is absorbed by stabilizer brush). Price: $250. Manufacturer: Shure Brothers, Inc., 222 Hartrey Ave., Evanston, IL 60204.

When we went on record several years ago as favoring the Shure V-15-IV phono cartridge, we opened a sluicegate of vitriol that has not been equalled since we dared to suggest, more recently, that the Sony PCM-1 digital recording system might have some merit.

I liked the V-15-IV in its G (spherical-tipped) version for several good reasons: Its unequalled tracking ability, which allowed it to sail effortlessly through such things as the Telarc 1812 cannon shots, its eminently smooth, sweet and musical high end, and its accuracy under the only valid listening conditions one can apply to a cartridge: Comparisons with a copy of the master tape that a disc was cut from.

Since that time, a magazine which -- Now get this! -- calls itself "Sensible Sound" described the V-15-IV as the "most accurate" cartridge available, then went on to declare a different one to be "the best" cartridge. This seemingly innocent little inconsistency is only part of a Grand Picture which is best described as the "anarchy of analog," about which I shall have a lot more to say in future, but not -- you will be pleased to note -- now.

Anyhow, Shure Brothers has now dared to publish chapter V of the V-15 saga. The V is superior to the IV in every objective respect: Trackability, distortion, high-frequency response, immunity to cable capacitance. Subjectively, I did not find it to be all that much better than the IV. It was better in one respect, not so good

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coils (from 500 milliHenries in the IV to 330 in the V), and by reducing the stylus tip mass so as to move the HF resonance up into the 35-kHz range. It should sound truly incredible, right? Right! It should. But it doesn't. It sounds very much like the V-15-IV (hyperelliptical), which we did not much care for because of its rather bright, hard quality. (Many audiophiles, in fact, judged our judgment on the basis of the hyperelliptical IV, either forgetting or overlooking the fact that it was the IVG -- the IV with the spherical tip -- which we were recommending.) In the case of the IV, that hardness tendency was solved by using the spherical tip. And while I have yet to try the V block, which plugs into the stylus receptacle, is used to check for proper torsional alignment, allowing the cartridge to be set perfectly horizontal to the disc surface (as viewed from in front) without risking damage to the stylus itself. Tangency is adjusted by means of a spacer strip with a jig that holds the cartridge precisely at right angles to the strip at the outermost of the standard tangency points, avoiding the need for eyeballing its degree of parallelism to printed lines -- the usual way of doing it. Tangency accuracy using this, with our SME 3009-III arm, was as good as we achieved with the Denneson Soundtracktor alignment device -- generally acknowledged to be the most accurate of any such devices available to the home user.

JGH

The ABX Comparator

Automatically pre-programmed A-B-test scorer. Price $395 for main Logic/Display module; $100 for RM-1 relay module. MANUFACTURER: ABX Company, Box 423, Troy, MI 48099

Now here's an inspired idea! A device which virtually guarantees the integrity of any comparative component-listening session.

The ABX comparator is a manually-operated switcher that allows a listener or group thereof to select between either of two input sources (A or B) or an unknown source (X). When you push the X button, the device switches to A or B, but it doesn't tell you which you're hearing. You're on your own. You make a note (on "score sheets" supplied) of whether you think it is A or B, and whether or not you prefer its sound, then go on to the next trial by pressing the button marked Up. This time, X may be the
same device as it was previously, or it may be the other device. Only the ABX knows which is which. If you wish to refresh your memory about the sound of A or B, just push the appropriate button. If you want to go back to a previous X, push the button marked Down as often as necessary to return to the desired test.

The device will accept up to 100 trials per run. When you're finished, you reset the device to 01 (first trial). The X button will light, as will the A or B button. The latter indication tells you which of the devices under test was component X for the first trial. Each time you press Up, the displayed number advances by one digit and the other readout tells you which component was X for that trial. You check off each "correct" answer against your score sheet and, when finished, you have a tally of the number of times you "guessed" which component was which. If you guessed right about 60% of the time, you've proven that you were hearing actual differences instead of just guessing. If you scored around 50%, you were deluding yourself. If you scored more than 60% wrong, you were undoubtedly hearing something, but you'd better not offer your services for a listening panel.

The sequence of X selections is pre-programmed into the comparator at the start of each set of trials; a different sequence is selected at random every time the device is turned on. The sequence for up to 100 trials is held in the device's memory for as long as it is left turned on. Pressing its Reset button does not clear the memory, as on many computers, but merely resets the counter to 01 (first trial), for verification of the preceding series of tests. There is even a clip on the back of the unit to hold 4 size-AA batteries, which will allow the ABX to hold its memory for upwards of 2 hours in case of a power failure or an accidental unplug-ging. You cannot however use the battery supply to read out your results, so if your power is not restored within a couple of hours, your tests go down the drain. Can't ask for everything.

Two remote devices are connected to the main ABX black box by 25-foot cables. One is the button module which allows you to select A, B or X for each trial. The other contains two relays plus 6 audio-cable receptacles. The relays connect one receptacle (per channel) to either of the other two receptacles, which means that the switching can feed one source (per channel) to either of two external-device inputs, or can feed two external-device outputs to a single (per channel) input.

The ABX cannot however switch a device's inputs and outputs simultaneously. If you are comparing the outputs from two devices, their inputs must remain in parallel across the source. This could be a liability under certain conditions, as when two power amplifiers present different load impedances to a preamplifier which does best with a fairly high load across its outputs. The amplifier which would normally sound best with that preamp will be handicapped by the reduction of its input impedance due to the paralleled impedance of the other amplifier, and NEITHER amp may sound as good with that preamp as it could. This may or may not be significant with any given pair of devices, but it is the kind of uncontrolled variable which tends to undermine the validity of any scientific experiment.

And speaking of power amplifiers, the ABX is not really designed for testing them at all. While the manual -- which itself is not as clear nor comprehensive as it could be -- shows how the relay box can be connected so as to switch loudspeakers between two
power amps, it cites the relays as being rated at 100 milliamps, which equates to a power-handling capacity of about .08 watts into an 8-ohm load. With most loudspeakers, this would elicit an output level of about 80 dB at 1 meter, which is not quite adequate for serious analytical listening. (Most audiophile systems are played at 90 to 100 dB on peaks.) Adding further to the difficulty of handling power amplifiers with the ABX is the fact that it has no level-set adjustments with which to match the gain of the amplifiers. Level matching is extremely important in A/B testing, so if the more-sensitive of the two power amplifiers does not have its own input level-set adjusts, these must be added externally. (They should be 100,000-ohm log potentiometers, 0.5 watt, located as close to the power amp as possible to minimize HF losses. These pots are available as "raw" parts from any electronics supply store, but they are not commercially available as preassembled devices from any source. I feel that ABX should at least make them available as an extra-cost option.)

Even with preamps (instead of power amps), there are a few problems which ABX should address themselves to. The lack of input switching means that one cannot do valid comparisons between preamps with a phono cartridge as the signal source. And since the biggest difference between the sounds of preamps is the way they handle impulse material from cartridges, it is essential for preamp evaluation that a phono source BE used. With the preamp inputs paralleled, the cartridge will see half of its recommended load impedance, resulting (usually) in rolled-off high end. Inserting a resistor in each channel to bring its load up to the requisite 47k ohms will reduce its output by 6 dB, and cranking up the gain on both preamps to compensate may make hum and/or hiss audible. And if one preamp happens to be a little noisier than the other, any listener can take the increase or decrease of noise as an unfair clue as to which preamp is currently occupying the X slot. (The best approach to preamp testing would be to lift the phono-input load resistors from one preamp, allowing the other to provide the load for both.)

As it now stands, the ABX is most easily used for comparisons between tape recorders, signal processors, and the high-level sections of preamplifiers. But that is not the end of the potential problems with this. Some devices, due to poor design, leaky capacitors or misadjustment, have small amounts of DC across their output or input connections. When one of these DC-offset connections is switched into the system, it will cause a click which, again, can be picked up by any perceptive listener as a clue to what component has just been switched in. High-value low-wattage resistors from the signal path to ground at all of the receptacles on the ABX will prevent this with most components, although NOT with power amplifiers. For valid comparisons of amps, any DC offset adjustments in them should first be trimmed to
eliminate every trace of switching clicks.

The only remaining question some purists may have about this is: Do its relay contacts affect the sound? I could not detect any sonic degradation whatsoever when I inserted the ABX between my Revox A-77 and the power amp it was feeding (sans preamp), but then it is impossible to make such a substitution instantaneously, for a direct comparison. So, I treated the ABX like a signal processor, and connected it into the Tape Monitor loop of my Berning TF-10 preamp. This allowed me to go either straight through the preamp, or to insert the ABX's switches into the signal path. I could not hear the SLIGHTEST difference, and I doubt that anyone else could under legitimate double-blind conditions. Of course, it could be argued that the Tape-switch contacts in the Berning were obscuring any differences due to the switches in the ABX, to which I reply, What system DOESN'T have any switching in the signal circuits? If yours allows a choice between phono and any other input source, it has a switch in it. And now that I've called attention to it, just watch for the crop of "dedicated" (i.e., single-input) preamps that will start to appear.

All cavils aside though, I feel that this is the first really significant contribution to the audio field in years -- not because it is a "breakthrough" in sound reproduction, but because it makes possible listening comparisons that the most hard-headed skeptics of subjective testing are going to have to take seriously.

In Volume 4, Number 9 of Stereophile, I observed that "subjective observations have no scientific respectability unless backed up by numbers." With the ABX comparator, we now have a way of gleaning those numbers. (See "As We See It" in this issue.)

I do not however feel that the present ABX Comparator is a "finished product." It leaves too many loose ends. The Logic and Display Module (the "heart" of the unit) plus the remote-control switcher ARE available without the relay box for $100, which allows one to put together any kind of relay switchers one wants -- with input/output switching and heavy-duty relays for loudspeaker switching. But I feel that ABX is only doing half their job by not making such proper switchers available themselves, at least as extra-cost options.

For now, though, I can only hope ABX sells a lot of these, for the more people (and audio groups) who own them, the more overwhelming will be the evidence that trained listeners CAN hear things that Julian Hirsch can't measure. But God, how I'd like to put some "underground" reporters' claims to fame to the test with this.

JGH

Acoustat TNT-200 Power Amplifier

Power rating 200 watts into 8 ohms; 300 watts into 4 ohms. Frequency response +0 to -3 dB from 2 to 500k Hz. Price $995. Manufacturer: Acoustat Corp., 3101 Southwest Terrace, Fort Lauderdale, FL 33315.

We heard this briefly about 3 months ago during a visit by Acoustat's Bob Rieman, and were immensely impressed, mainly by its remarkably sweet yet detailed high end, but also by its ability to control what has proven to be the Acoustat Model Four's only real liability: A tendency towards heaviness in the middle-bass and low end in a typically-sized listening room. We were so pleased with what we heard, and Bob was so pleased that we were pleased, that he promised to send us one for an
extended listen as soon as he returned to Florida. Several weeks and at least as many phone calls later, we finally pried a TNT-200 loose from Acoustat.

In the model number, TNT would seem to stand for Trans-Nova-Twin, but they aren't kidding anyone. Obviously, what the TNT REALLY means is Tri-Nitro-Toluene, and THAT is dynamite! At 200 watts per channel into 8 ohms, this isn't exactly a weakling. But dynamite? Not quite.

But how does TNT translate into terms of listening quality? I shall quote for a moment from Acoustat's blurb sheet: "Say the tube admirers, 'Solid-state amplifiers (using bipolar transistor technology) have an 'electronic' sound, particularly in the critical midrange where most of the musical information is contained. This glare or edginess is never heard in live music.'"

"Reply the solid-state advocates: 'Electronic glare? Perhaps, but that's no reason to go back to 75-year-old tube technology with its audibly slower transient response and soft bottom-and top-end performance (due partly to the necessity for an output transformer).''

The rest of Acoustat's blurb explains why all of the problems of solid state have been solved by using FETs instead of bipolar transistors. Unfortunately, even in 1982, there ARE no audio panaceas. We will readily grant Acoustat their point about the "soft" bottom of most tubed amplifiers, but we have never yet had an amplifier in the house that could surpass the high end of the Berning EA-230 feeding electrostats.

While I'm being disagreeable, I will take issue with another small detail in Acoustat's literature. As the person who coined the term "glare" to describe a certain kind of sound, I feel that I am on firm ground in telling Acoustat that the word does NOT apply to most solid-state amplifiers. "Glare" is spurious brightness -- an apparent exaggeration of the upper-middle range -- and is more often a characteristic of TUBED electronics than transistors. Typical solid-state amplifiers, on the other hand, are usually somewhat laid-back-sounding because of an apparent DEFICIENCY in this range. Which of these is the more ACCURATE is something that is indefinitely arguable but is hardly relevant here, because the tube adherent's major criticism of the so-called solid-state componentry is its dry (chalky-textured) or gritty high end, which is not related to brightness at all.

When We finally got around to firing up the TNT-200, it didn't impress us as it had on first listen. It was generally very good, but lacked the depth and inner detailing of the Berning, and I couldn't even get ecstatic over its high end, which was slightly dry and -- Heaven forbid, after Acoustat's literature -- "mechanical." It did not sound like the same amplifier I had heard before, except for one minor detail. It still controlled the Acoustat Four better than anything we had tried on those speakers.

Then, barely a week after I had installed the TNT-200, I received a small brown-paper envelope from Acoustat containing, guess what? Parts and instructions for a modification which would "dramatically" improve the unit's performance. Now let me tell you how I feel about doing modifications. During my 40-odd years in audio (some of which were VERY odd), I have constructed, serviced and modified more components than I care to think about. As of now, I would be perfectly content never to smell burning rosin again. So when I received Acoustat's little Care package, my first reaction was to send it and the amplifier back with a note saying "Do it yourself and return it." Then I
remembered how much trouble I had had getting the amplifier in the first place, and decided to do it myself.

The modification involved nothing more than the replacement of four small resistors with new ones supplied -- a snap for any experienced solderer with flawless near-vision except when, as in my case, one resistor wire (the last one, as usual) refuses to pull out of its hole in the PC board. The problem was solved by attaching one side of that resistor to the wire next to it (which, conveniently, went to the same place on the printed circuit board.) The job then requires the use of a sensitive DC volt meter for final adjustments of both channels. In short, this is one mod I would recommend having your dealer do, unless you are well into the nuts-and-bolts (and solder) aspects of audio. But it IS a modification that I would recommend.

I'm not sure I would go so far as to say that the improvement in sound was "dramatic," but it was certainly audible and more than worth the effort. The high end improved both in detail and in sweetness, definition across the board became perceptibly better, and even the low end seemed to be somewhat better-defined. Feeding the Acoustat Fours, the TNT-200 was substantially superior in every respect to our old faithful Infinity HCA, and approached but did not equal the fantastic (and fantastically high-priced) Sony Esprit TA-N900 when driving Infinity RS-4.5A speakers. And it was with these that we first noticed what appears to be a conditional liability with the TNT-200. It seems to dry up the low end of some dynamic woofers.

We suspect that this might be due to the Acoustat amplifier's extremely high damping factor. Many dynamic woofers are designed with somewhat less than critical damping, so their slight hangover gives an impression of slightly added fullness at the low end. The TNT-2000 puts the brakes on this slight resonance, and the result with most dynamic speakers is a tautness and dryness at the low end which can on occasion border on outright thinness. Electrostats are less affected because much of their damping is provided by the air loading on their panels. The "braking" effect of amplifier damping on them is consequently less. The tendency to lean-out the low end remains however, and must be considered when mating this amplifier to ANY speaker system. Most large full-range electrostats, such as Acoustat's wide (3-panel and more) systems and the large Sound Labs systems, have that tendency to sound overly fat in the mid-bass when used in average-sized rooms, and for those, the TNT-200 is ideal. Many large dynamic systems too can benefit from what this amplifier does, for what it does is suppress the mid-bass boom while adding detail to the low-bass range (without significantly effecting low-bass output).

In very large rooms, the TNT-200 may overdo this low-end clamping, making the speakers sound somewhat thin and bright. It may also have the same effect in a typical room when the speakers are inherently well-balanced to begin with. The Acoustat Two and Three, for example, will often provide superb over-all balance in a room of moderate size when driven by, say, a pair of strapped Berning EA-235s. Substituting the TNT-200 here is likely to spoil that balance, making the system bass-shy. So, how this will sound with a given speaker system depends to a great extent on how it sounds with other amplifiers. If there seems to be a need for controlling low end, the TNT-200 is ideal. Otherwise, it may not be the best thing you can use. Except for that one cautionary suggestion, I can rec-
ommend this virtually without qual-
ification. Its high end, on even
the most revealing loudspeakers,
is almost without peer, and in all
other respects it intrudes on the
music about as little as anything
I have heard.

And on a watts-per-dollar ba-
sis, this has to be one the best
buys you can make right now. JGH

B&O Beosystem 8000

Can a perfectionist find happi-
ness with a complete audio system
that costs $6000, does practically
everything one could ask, automa-
tically or by remote control, and
has won international acclaim plus
a few awards for its styling? A
lot of readers have apparently
been eying all those goodies with
interest and wondering just how
much audio quality that much con-
venience would cost them. We
shall endeavor to answer that.

The Beosystem 8000 has been
written up in so many other maga-
zines that it would be pointless
for us to do yet another lengthy
treatise on its multitudinous vir-
tues. We won't reiterate them in
the usual detail, simply because
they have been said so many times.
We will, instead, take a close
look at its sonic attributes,
applying audiophile standards ra-
ther than those of a mass-market
audio publication. Because we
will be looking for weaknesses
rather than strong points, this
may read like a rather negative
report. It is not intended to be.
The Beosystem 8000 is a triumph of
design, a joy to use, and — like
a great work of art — a continu-
ing pleasure to look at. But we
chose to approach this report from
an audiophile's viewpoint, by eva-
luating the bottom line, so to
speak: its sound. And that is
where it falls short of ultimate
grace.

To begin with, the system lacks
several refinements which audi-
ophiles today consider to be of
importance. It uses what appears
to be common lamp cord (estimated
at 18 gauge) for the speaker
wires, the wires are not long
enough to put the speakers any-
wheres but along the same wall as
the rest of the system, and the
turntable "mat" is no mat at all,

The Beogram automatic turntable.
but consists of a series of radial
ridges of hard plastic. Not ex-
actly a purist approach.

The speaker wires are not easi-
ly lengthened because there are
special molded plugs at each end,
and while extras are supplied for
the amplifier end, no such are
supplied for the speaker end.
Lengthening the cables, then,
would involve cutting them and
inserting additional lengths into
the cables supplied. But the
limited speaker mobility is not
really a liability because of the
remote control, which allows you
to control most of the system
functions from anywhere in the
room.

Neither are the speaker cables
themselves all that much of a
shortcoming, as the speakers are
probably the weakest link in this
entire system. These are 4-way
units with the drivers mounted on
a shallow concave baffle that
gives the impression that the
speakers are slightly stoop-shoul-
dered. We do not understand the
logic of this arrangement, and can
only report that it does some odd
things to the vertical dispersion.
The brightness of the system varies widely depending on one's angle above the axis of the lower-middle-range speaker. From on-axis to about 5 degrees above that axis, there is a distinct suckout in the brightness region, which diminishes as the angle increases. Unless you are sitting high and close, you'll hear more suckout than not. There is a tilt adjust at the rear of each speaker, but its range is inadequate to correct for this apparent phasing anomaly. What it boils down to is that the speakers sound better when you stand than when you sit. Most of us prefer to listen while seated.

Otherwise, the speakers have very respectable deep-bass range, but the bass is neither very taut nor very detailed. With our ears at a typical seated-listener height, the over-all sound was rather heavy, super-rich, laid-back and veiled, with little inner detailing and a markedly closed-in top. Paradoxically, there was also a tendency towards hardness when reproducing massed violins at anything above moderate listening levels (that is, 85 dB and above). At lower levels, instrumental timbres were somewhat darkened. Stereo imaging was very good although somewhat "phasey" at times. (The term "phasey" refers to an odd sensation of pressure which is felt in the ears rather than heard, when large segments of the audio spectrum are reaching the ears out of phase while others are in-phase.) Front-to-back perspective was fairly well reproduced.

The sound from discs and tapes was remarkably similar, with both reflecting, basically, the sound of the loudspeakers. B&O's tiny, featherweight cartridge proved to have quite remarkable tracking ability, handling all the mid-range and high-end modulation we could feed it, and pooping out only on the horrendous cannon blasts on Telarc's 1812 Overture, half of which caused it to skip a groove or two. (That 1812 is kind of a lackluster performance, but all audiophiles should own it anyway because its infra-bass cannon thuds, if you can track them, are an almost-unique experience! These are what critics have in mind when they speak of visceral impact.) The pickup's very low mass, which is what makes it unable to trace heavy deep-bass material, also makes it able to trace rapid record warps that would cause any audiophile pickup to put out severe subsonics or to hop out of the groove.

By itself, feeding our reference system of the moment, the phono unit had a slightly dark quality, reflecting a very gradual measured downward tilt with rising frequency. Over-all, the cartridge had a sweet but somewhat veiled sound, with rounded-off transients and, puzzlingly, a slight tendency to exaggerate surface noise. High-level inner-groove passages elicited a certain feeling of insecurity, but outright mistracking was rarely audible as such.
The turntable's suspension was very effective against floorborne jounces and feedback, but susceptibility to airborne feedback was somewhat greater than average — a result, I would guess, of the almost-total lack of disc damping (because of the ribs instead of a proper platter mat). And I suspect that some of the phono unit's lack of detail was also traceable to the same thing, because tapping the disc surface produced the same kind of "Bunk!" that one hears when other turntables are used without any mat at all. (With a well-damped disc, a fingernail tap produces a very brief "Bup!") This is not easily remedied here, because if you simply lay on a better mat, it lifts the disc so high that the raised arm won't clear it. And even if it did, the new mat would prevent the B&O's optical sensing system from working (because this reads the contrast between the reflectance of the metal platter and the black of the disc). The 'table would simply not function properly. So, we are stuck with the raised ridges.

In audiophile terms, the cassette machine proved to be the best component of the lot. With excellent speed regulation and very low noise, its only sonic flaws were a subtle loss of deep bass and a closed-in high end, both of which were clearly evident on the frequency-response measurements. We have since learned however that the Model 8000 recorder we tested was replaced with a model 8002, which we did not test. It is probably safe to assume however that the new one will be at least as good as the 8000.

The tuner section of the main receiver was surprisingly good, with very low distortion, excellent quieting, and enough sensitivity to provide satisfactory listening from all but the most borderline fringe stations. The audio section was very good, but little better than any number of other high-priced receivers. Its sound was a little on the lean side, but otherwise, there was little about its sound to comment on. Everything about it was very good, nothing was superb.

So, it worth the money? For sound alone, you can do better for less money. A system comprised of our Class-C Recommended Components will give higher accuracy for several hundred dollars less, but (con't on page 27)
**Recordings**

**AN OLD VIENNESE SCHRAMMEL CONCERT.** Sonic Arts Digital Recording. Sonic Arts Laboratory Series 17.

This is a delightful recording of some not-very-serious or intense music. Just music to relax, drink, and converse by. The title is a generic term derived from the name of a composer, Johann Schrammel, who played with his brother at Grinzing, a suburb of Vienna, known for its many watering holes where the Viennese would go to enjoy a glass of wine and the latest popular music.

Some of these charming tunes are reproduced on this disc. The arrangements are similar to those one might have heard in the early 1900's, deploying violins, accordion, and bass. The recording is excellent although I must admit that the demands of four instruments on miking and positioning are not all that stringent. (This month's reviews all seem to be going that way.) However, capturing the authentic atmosphere and the proper sonorities is a more difficult accomplishment, which Mr. Gar Kulka and his crew have done to perfection.

This is a record to play with good friends, good wine, and good conversation.


As usual, this is a superlative recording. In fact, it is rare these days for Sheffield to put out a less than technically superb disc. Although the reproduction demands of violin and piano are not all that stringent in terms of power and frequency range, they are sufficiently rough in terms of tonal structure that they rarely reproduce naturally. With this, I felt as if the performers were playing in my home.

The program is most enjoyable. Romantic chamber music repertoire is challenging as well as exciting to listen to. These two sonatas are no exception. The Strauss is early, written when he was about twenty three, and exhibits some of the strongly personal traits which mark his more-mature works. The Dvorak is a mature work dating from his mid-40s. It is a lyrical as well as emphatic work with a wonderful use of double stops for the violin in the second movement. There are hints of Dvorak's Slavic heritage throughout. Most enjoyable, as is the more dramatic and darkly colored Strauss.

Despite the fact that this direct-to-disc involves only two instruments, and is thus hardly what one could consider a system showoff record, it must be considered a Top-of-the-Pile recording.


It is hard for me to be objective about a record such as this. My very being responds to it, not only to the music but to the ideas and feelings behind it. Fortunately for me, this happens to be an excellent recording, with some extraordinary low end on it, so I need not compromise either my...
critical faculties or my sentiments.

For those of you who may not be familiar with Paul Winter's earlier recording, "Common Ground," Mr. Winter has become deeply involved with animals. It began when he started playing his saxophone to wolves, who responded to his music, at times echoing the sounds. He developed this into "Common Ground," which was dedicated to the lords of the animal kingdom: whale, wolf, and eagle. "Callings" is the first album in a projected trilogy, devoted to these three powerful creatures. Each album will be dedicated to the ruler of one of the three environs: air, sea, and earth. "Callings" was inspired by a sea lion pup who joined Mr. Winter and some friends when they were camping in Baja California and shared an evening with them. They called her Silkie. Although the album is primarily devoted to mammals of the sea -- whales, dolphins, walruses and seals -- it also includes sounds of wolves and polar bears.

The animal sounds are incorporated into the musical context, where they seem as natural as any instrument in the scoring. The music itself can be awesome, as in "Sea Storm" (which includes a clap of thunder that seemed almost a divine embellishment of the music) and "Blues Cathedral," which contains some very deep sounds -- the voices of Blue Whales raised in frequency by two and four times. Despite this, they are still pitched so low as to be below the bottom range of many good speaker systems.

The music can also be very gentle, loving and wistful, as in "Dance of The Silkies," as well as lyrical as in "Seal Eyes" and "Sea Joy," the latter a gentle dance tune. The varying combinations of sax, oboe, English horn, cello, harp and several percussion instruments too numerous to list, afford a wide variety of moods and textures.

This record may not be everyone's cup of tea, but for those who have an affinity for animals, and those who have enjoyed Paul Winter's earlier records, it will prove a delight.

The recording itself is well made and mixed. It was digitally mastered on the 3M System, about which the least I can say is that it doesn't sound digital. All in all, this is a most amazing recording and one which will interest as well as delight many record collectors.


This is an outstanding analog recording. The mike perspective is perfect, and the rich sounds of the stringed instruments are captured faithfully.

The performance of this first version of Opus 18, No. 1 is elegant. Tempi are totally appropriate to the various movements and the playing is excellent.

Like the Sheffield sonata recording reviewed in this issue, this isn't ardent-audophile fare due to the fact that there is no deep bass, nor is the music very demanding of the speakers as the group is limited in size. There are high frequencies from the violins which will test your system's treble response. It is a delightful reproduction of chamber music, especially a relatively unknown version of a familiar work. Close scrutiny of the score or careful listening to the two versions of this work will reveal the developing musical maturity of the composer. The slight changes from one score to the other add strength to the quartet.

All in all I think, from a musical as well as a recording standpoint, this belongs in any serious collector's library.
LARRY POTINE AND HIS BIG BIG BAND. Repertoire Records. RR0178.

This big band sounds more like the bands with which I remember. It takes me back to the magic days of radio when, as a very small child, I was exposed to "Make-Believe Ballroom", the music from the Starlight Roof of the Waldorf Astoria and other unattainable musical delights. But, isn't this the purpose of any record which is listed as being produced by Nostalgia/Repertoire Recordings?

Although the instruments here are not as closely-miked as in the RealTime Don Menza, the acoustics are drier and thus much more evocative of the recorded sound of the 40s. There is a lovely brassy quality to the sound, and the brasses are emphasized in the arrangements. Despite the fact that this is not specifically aimed at audiophiles (i.e. not digitally recorded nor direct to disc), it is nonetheless an excellent recording, although its appeal is likely to be primarily to those who remember When.

DON MENZA AND HIS '80S BIG BAND. RealTime Records. Digital Recording. RT 301.

This is the big band sound which was prevalent during the last few years of the big bands. It is more of an amalgam between jazz and the big band sound than anything else. It is born of Guy Lombardo, Tommy Dorsey, Sauter-Finnegan, and Dave Brubeck. Compared to the sounds of Guy Lombardo and his ilk, the sound is more textured and the musical structure more involved. For sentimental reasons I prefer the older, more fortys music. This recording has much to recommend it nonetheless. There is a good use of prominent brasses especially during the selections "Burnin" and "Dizzyland" which are well recorded. The recording itself is good. Miking is well done, and there are no annoyances either in the treble or the bass. For Big Band aficionados a good addition to the collection.

The Digital Dirties Explained?!

A conversation with Sonic Arts President Leo de Gar Kulka at the Chicago CBS turned up what just might be the answer to some of the digital-disc dirties.

Most users of digital tape recording equipment report that the sound of playbacks is incredibly clean. Yet analog discs mastered from digital tapes nearly always have an unpleasantly hard, gritty high end. Mr. Gar Kulka suggested to us a possible reason.

He happened to notice some scanning-electron-microscope photos of Telarc record grooves in a recent issue of "Audio Alternatives" magazine, and observed a very odd shape to the modulations. They seemed to be comprised of triangular waves rather than the usual rounded modulations of music waveforms.

Now typically, a triangular wave is what is cut into a disc when an unequalized square wave is fed to the disc cutter. A sine wave, incrementing through such triangular steps instead of a smooth undulation, would represent a series of square-topped steps -- which is exactly what we get when it goes through the quantization phase of digital processing.

Most digital recorders have extremely thorough filtering of their output, to eliminate these steps by averaging the energy between each one. How then could these apparent quantization signals get onto a digital disc? Mr. Gar Kulka theorized that they
might come from a digital delay device.

In order to efficiently utilize the space on a disc, it is customary to vary both the pitch (spacing) and depth (width) of the groove while cutting, reducing both during quiet musical passages and increasing both during loud passages. When a loud passage follows a quiet one, the grooves must be spaced out far enough in advance that at least one disc revolution takes place before the loud signal occurs. If this is not done, the grooves will not be far enough apart when the loud modulation comes, and the loud one may cut across into the adjacent softly-modulated groove. With modern disc-cutting systems, both adjustments are made automatically, with the control signal coming from an advance head located about 22 inches ahead of the analog recorder's playback head. At a tape speed of 15 ips, this gives about 1-1/2 seconds between the groove adjustment and the sound that calls for it.

This evidently cannot be done with any of the existing digital tape machines, because the audio signal is not reconstituted until it reaches the main playback head. If it is desired to use automatic pitch and depth control, the only simple way to do it is to use the main digital output signal as the control for the P&D system, and feed the cutterhead's signal through a delay device to provide the necessary one-and-a-half-second time difference between the control signal and the cutterhead signal. And most digital delay devices do not have nearly as effective filtering at their outputs as do digital recorders.

A triangular-wave modulation at the high frequency at which delay devices sample (around 20 kHz) is completely untrackable. The result would HAVE to be a dirty high end, whose severity would be worst with high-resolution styli like line-contacts, and least with sphericals, which would tend to average out the shoulders of the triangular waves.

This model of what ails digitally-mastered discs sounded worthy of investigation, which was what we made. What we found weakened the case but did not kill it. We talked first to Telarc's Bob Renner about this, and then to Stan Ricker, who used to cut Telarc's discs when they were being done by the JVC cutting center. Both gave us the same answer: No Telarc disc has ever been cut using a digital delay to provide an advance signal for pitch and depth control. The Soundstream system, it seems, is 4-channel, and the extra two channels have always been used to provide the necessary (for automatic groove control) advance signal. Digital recordings have absolute long-term timing accuracy, so it is simple enough to make a copy from the original digital master and dub this onto the two leftover tracks with the requisite amount of advance time to provide proper control for the cutterhead.

So as far as Telarc's discs are concerned, this would seem to be a dead end. But we don't think it is a dead issue by any means. Cutting information is harder to get from larger record companies and, although we have not as yet tried to pursue this with Angel, Vanguard, RCA and CBS, we are not too confident of getting answers from those sources. Our investigations to date have however turned up some discouraging news for digiphobes: It seems that a number of record companies, evidently oblivious to the possible sonic results thereof, are using digital delays routinely to provide advance control signals for ALL tapes, digitally OR analog-mastered. Which means that the analog-mastered tapes are digital by the time they get onto the disc. A few have apparently realized the
error of that way and have quit, but not before cutting a number of discs that are at your friendly local record store now. Others are still at it, and may or may not catch on within the foreseeable future. So if you come across an analog-mastered disc that sounds like a bad digitally-mastered one, don't be too confused by what you hear. You may be right.

Interestingly, none of Sonic Arts' digitally-mastered discs sound like digitalis, and all had their pitch and depth MANUALLY controlled.

Coming Up

Received for testing since Summer CES are the Pink Triangle and CW&J Walker turntable, Audio Interface's new head amp, the Astatic MF-100 cartridge, and the Orpheus S505, Black Acoustics Rainbow, and Scandinavian Sounds QLN speakers. More are expected, including a Sony PCM-F1 digital tape system and a pair of Pressure-Zone (PZM) microphones.

The Ultrasonic Shure

A couple of issues back, we published an item entitled "Test Records and How to Misinterpret Them," which explained how running them at speeds other than the nominal 33.3 rpm could reveal whether observed response deviations are coming from the cartridge under test or from the test record itself. Our experience with the new Shure V-15-V provides a first-hand example of how this works.

For measuring frequency response above 20,000 Hz, we use CBS Labs' STR-120, which has sweep tones on it from 500 to 50,000 Hz. Using this with the V-15-V, we found a rather narrow 3-dB peak in the right channel, centered around 30 kHz. The response above that rolled off rapidly, measuring 3 dB down at 37 kHz. Question: Was the peak in the cartridge or the test record? And was 37kHz really the cartridge's upper limit? Running the disc at 45 rpm provided a quick answer to both questions.

For all of our response measurements, we use the Neutrik Audiotracer -- a thermal-writing "pen" recorder which gives automatic readouts of frequency response when its own built-in oscillator is used. The oscillator frequency, from 20 to 20,000 Hz or from 200 to 200,000 Hz, is determined by rotation of the feed system which advances the pre-printed graph paper. (A switch selects the range of frequencies.) The roller which controls the oscillator frequency is friction-coupled to the paper-feed system, so that when the paper is set to start advancing from 20 Hz, the oscillator can be set to start at the same frequency. With the drive clutch engaged, the frequency calibrations on the paper will then "track" with those of the oscillator regardless of the speed at which the paper advances. This neat arrangement goes down the sluice when an EXTERNAL sweep is used, as when the signal source is a test record instead of the Audiotracer's own oscillator. This situation requires that the paper-advance speed be very accurately set, so that a sweep from, say, 40 to 20,000 Hz starts at the 40-Hz mark on the printed graph and ends up at the 20,000 mark at the completion of the sweep.

This is further complicated when the sweep-frequency record is run at a speed other than 33.3 rpm. At 45 rpm, for example, everything is speeded up by a factor of 1.35. All frequencies in the sweep are transposed upwards by 1.35 times, and the duration of the sweep is only 0.74 of what it is at 33.3. If the Audiotracer's speed is not adjusted for
this, the resulting curve will be horizontally compressed, and its frequencies will not "track" with the calibrations on the graph. Since the speed of the sweep has been increased by 1.35 times, it is necessary to increase the speed of the paper feed by that factor in order for the 45-rpm sweep to be comparable with the 33.3-rpm one on the same graph. Since the paper-speed adjust on the AudioTracer is quite coarse (its only major shortcoming), accurate setting can only be done by trial and error.

On the STR-120, the usual sweep starts at 500 Hz and ends at 50 kHz. The pre-printed graphs on the AudioTracer paper span 20 to 20,000 Hz, so in order to prevent this 500-to-50,000-Hz sweep from going off the end of the graph, it is necessary to start recording with the "pen" at the 50-Hz calibration instead of at 500. With the paper feeding at the proper speed, the end of the sweep (50 kHz) will then correspond to the 5000 Hz calibration on the graph.

Since running the disc at 45 rpm transposes all frequencies upwards by 1.35 times, this frequency run must start with the "pen" at the 675-Hz calibration mark, and the paper speed must be increased so that, during the shorter duration of the sweep, the graph is still advanced from that mark to a frequency of 10 times the starting frequency: 6,750 Hz. When this adjustment is made, it is possible to superimpose the two resulting curves and to draw some conclusions from them.

The curves shown below are laterally displaced so that the frequencies along each one coincide. Curve A is the 500-to-50,000-Hz sweep at 33.3 rpm, and curve B is the 675-to-67,500-Hz sweep at 45 rpm. Note that the response peak in both curves occurs at exactly the same frequency. If the peak had been in the test record, it would have moved upwards in frequency when the disc was speeded up. The fact that it stayed put is proof that the peak is in the cartridge, not the record. Note also that, above the peak, the rate at which the measured output falls off is different for the two curves. This would indicate that the attenuation is a function of both the cartridge AND the test record, which means that it is not possible to cite a precise figure for the cartridge's high-end limit, but merely to state that its output falls off rapidly beyond 35 to 40 kHz. That would be pretty good for a moving-coil cartridge. For a moving-magnet cartridge it is extraordinary!

(It should be noted that, while MC cartridges do not lose high-frequency response as a result of the inductance of their coils, as

The V-15-V's measured response from the STR-120 record. A is at 33.3 rpm and the frequency span is 500 to 50,000 Hz. B is at 45 rpm, for 765 to 67.5kHz.
do most moving-magnet types, many of them have such high moving mass in the stylus/armature/coil assembly that they exhibit a rapid falloff in high end above a certain frequency because of the stylus' inability to respond to extremely rapid modulations of the groove. Such MC cartridges will actually exhibit poorer transient response than MM cartridges whose sine-wave response extends out to a substantially higher frequency.)

LAUREL RECORD COMPANY

Laurel Record Company is a new but prestigious label for fine recordings. Although their repertoire is definitely not audiophile-oriented, the sonic quality of their recordings should disappoint no one.

The first Laurel release we received for review is a Beethoven Trio, but Herschel Gilbert, President of the Company, assures me that this is an anomoly. "What point is there in releasing the fourteenth recording of a Beethoven Quartet or the tenth recording of a Mozart Concerto?" he says. Instead, his company is concentrating on contemporary American composers with, he states, more discrimination than that of CRI.

A graduate of Juilliard, Mr. Gilbert has been a composer of film and television scores for more than 40 years. He believes devoutly that people need to hear the music that is being written now, adding that they will never learn whether or not they like it unless the DO hear it. Mr. Gilbert uses Neumann SM-2 intensity stereo microphones for his recordings, and he assures me that once the balances are set, "I do not diddle". His opinions of some of the more highly-touted audiophile devices are of interest, as they arise from a completely musical background rather than the perspective of an audio engineer.

For example he feels that Soundstream has a problem with their midhighs as they do not use enough bits for their quantization. The Dolby devices are only useful for cassettes as they tend to thin out the strings, and they bother the trumpets as well as other brasses.

One aspect of his records of which he is very proud is the pressings. He suggests holding a Laurel recording to a bright light and looking through it. Then compare it with a Columbia (Heaven forbid!) or an RCA. You will immediately notice a difference. He is using a new type of vinyl which is being produced in California. He is proud of both the qualities of the discs and the quality of musicianship and program material which is presented on them. Surely such a company deserves our support. MG

(con't from page 9)

were more subtle and gave less to choose between. In the ISOS demonstration against the Linn the differences were much more dramatic -- enough to make you suspect the test was rigged somehow. The apparent volume level when the record was played on the ISOS was significantly higher; only by careful concentration could one hear that the softest passages were indeed at the same level, and that therefore there was no hidden resistor in one of the circuits. This was the most startling, and suspicious, characteristic. The other very noticeable outcome of the test, not suspicious at all, was significantly better revelation of detail on the ISOS. Using an Ortofon test record of hand clapping it was possible to hear far more of the subtleties of the (numbers of) hands clapping, their little overlaps, with the ISOS. The Linn, while no slouch at the test (the overall sound, from both 'tables, was excellent), simply didn't offer up as much of that interesting detail. The ISOS
people claim "tighter (faster), deeper bass" but I was unable to confirm this. In any case, the demonstration showed that the ISOS is one to be reckoned with, and we hope to see one as soon as it's out of the prototype stage (supposedly August or September, but you know how that goes). By the way, I have no reason to question the integrity of the ISOS people or their demonstration; I only mention my suspicions because differences between components are not normally that dramatic and our subjective immediate response is to say, "Now, wait a minute . . ." It does make me want to try the test myself. 

LA

(con't from page 19)

it won't do anything automatically or by remote control, and it certainly won't look as nice. All in all, we would say that this system IS worth the money if its other, non-sonic attributes are worth anything to you. But like any other system where cost is any consideration at all, this one embodies some compromises. Only you can decide how much they will bother you. JGH

Time Up?

If your address label on this issue bears an alphanumeric code of V-5, this is the last issue of your subscription. If you have just renewed, ignore this and accept our thanks. If you haven't, now is the time to do it. We're publishing so often now that a delay of only a week in renewing your sub will miss you an issue. Take advantage of the business reply cards we're now supplying with each magazine and send in your renewal to avoid delay in receipt of V-6.

Reprints and Back Issues

We still have on hand a limited number of the softbound reprints of our first and second 12 issues. If you're curious about how and why this whole business of perfectionist audio got started (It was all our fault!), these two reprints are a not-too-compact history of the whole sordid affair.

Volume 1 contains 240 pages and covers the years 1962 to 1966, and Volume 2 has 290 pages and covers up to Spring of 1971. Both are 8-1/2 by 11 size (That's the way we were then!), and the price is $25 each.

Also available in rapidly-dwindling quantities, for the princely sum of $4 each, are original, unsullied copies of the following back issues (Volume and Number): 3-3, 3-4, 3-5, 3-6, 3-7, 3-9, 3-11, 3-12, 4-1, 4-5, 4-6, 4-8, 4-10, 5-1 and 5-2. Good Xerox copies of out-of-print issues can be obtained at varying prices, depending on number of pages. Inquire about specific issues.

The Business Reply envelope has spaces for ordering back issues, as well as a space for requesting a synopsis of the back issues so you'll know what you're ordering.
The Literature

Hearing vs Specifications

Perhaps we are a little odd in holding to this view, but we have always felt that, the larger a magazine's circulation, the more responsibility it should feel for editorial accuracy. The success, and content, of popweeklies like "World" and "The National Enquirer" are proof that not everyone agrees with us, but we have come to expect better from "Stereo Review" than their recent (May '82) article by Mark Davis on "Audio Specifications and Human Hearing."

The article was generally excellent, offering explanations for several things (such as our ability to perceive vertical direction as well as lateral) which are not well understood by most audiophiles but should be. Our main criticism of the piece is that it approached some points backwards, as it were. Conclusions about what we can and cannot hear were based upon present knowledge about audio measurements and hearing, rather than upon direct observations about what we CAN in fact perceive. As a result, Mr. Davis ended up telling SR's readers that it is not possible to hear certain things that a substantial number of them are in fact hearing. For example...

Through several steps of irrefutable logic, Mr. Davis concludes that "frequency response -- can be considered perceptually 'flat' if, after one-third-octave, third-order smoothing. the response deviations do not exceed 1 dB (0.3 dB if you're really fussy)." Note that, to be "really fussy," this means a total allowable deviation of 0.6 dB. Some critical listeners can detect a broadband deviation of less than 0.1 dB; most of us can readily hear 0.6 dB. More about this subsequently, though.

Many perfectionists will also take issue with Mr. Davis's assertion that phase shift is inaudible until it reaches disastrous proportions, but we should point out here that the audibility of phase shift has NOT as yet been positively confirmed.* Enough people are however concerned about it to warrant more than a summary dismissal.

Mr. Davis states that "as far as human hearing is concerned, transient response is a red herring." This may in fact be true of human hearing, but it is NOT true of reproducing equipment. Many components exhibit asymmetrical behavior in the presence of transients, having unequal rise and fall times and, often, unequal amplitudes on positive and negative half-cycles. Resonant ringing, too, is often induced only by transient material. These transient aberrations are easily measurable, as is (with somewhat greater difficulty) transient intermodulation distortion. The audibility of these things has never been con-

*It is necessary here to distinguish between phase shift -- a relative delay between two signals reaching the ears -- and phasing interference which causes destructive cancellation between two signals. The latter, which involves phase shift between two sources radiating a common range of frequencies, causes pronounced dips in the frequency response of their combined output. This is the mechanism which produces the well-known "vertical-venetian-blind" effect from certain loudspeakers. It takes only a few degrees of phase shift between drivers to cause an audible response dip.
clusively proven, but what HAS been demonstrated time and again is that, when these things are minimized, the equipment invariably sounds better -- more musical, lucid and detailed. Not conclusive proof perhaps, but again, there is too much evidence to support the audibility of transient performance to simply dismiss the whole thing as a "red herring."

Mr. Davis also falls into the common trap of assuming that "the spectral complexity of most music and speech will do a much better job of masking distortion than a pure tone." This is probably true to an extent when the signal source is a very clean source such as an original master tape, but it is far from true when the signal is -- much more typically -- from a phono disc. This in fact what burns us up about so many of this kind of debunking article: The authors pursue their own line of reasoned thought without apparent knowledge or consideration of what has been published elsewhere which might tend to undermine their conclusions.

So, we must reiterate yet again the point which is so often overlooked in these discussions about what is and is not audible. No stylus ever tracks the groove perfectly. At all recorded levels above moderate, the stylus loses contact momentarily first with one groove wall, then with the other, and as level increases, discontinuities become more frequent and more severe. Each time the lost contact is regained, the stylus collides with the groove wall, producing a click. The energy in the click signal spans a wide range from several thousand kilo-Hertz up to the electrical high-end limit of the cartridge (as distinguished from its mechanical limit, which is a function of stylus mass). In the case of many moving-coil types, the electrical limit can extend strongly out to beyond 60 kHz.

These transient pulses hit the preamplifier with precisely the kind of signals it has the toughest time coping with: High-energy transients. If it can be overloaded, if it has asymmetrical performance, if it has TIM or if it tends to ring, these mistracking-induced transients will cause it to do so. Any electrical distortion in the preamp -- even when outright overload does not take place -- is sufficient to cause intermodulation between the mistracking transients and the spurious upper harmonics of the recorded program material to generate sum-and-difference products the latter of which become splattered down into the upper part of the audible range. The result is an irritating "texturing" of the sound which, depending upon the amount of "inaudible" high-end imperfections in the preamp, can range all the way from a subtle dryness or "chalky" treble quality to a harsh, gritty edge.

Further proof that this phenomenon exists is the fact that sound from discs is affected by the way components behave in the obviously-inaudible frequency range above 100 kHz and as far out as 1 MHz! Case in point: Audio interconnects and loudspeaker cables, which exhibit no measurable differences at all below 30 kHz but increasing differences above, can nonetheless have a marked effect on the quality of the upper part of the audio range. There is NO WAY a musical sound, recorded with mikes having a 20-kHz upper limit, can provide information in the 1-mHz region except as a result of distortion, and no way these distortion products can be audible unless they are producing spurious difference products.

**AUDIO VERITY:**
The better the system, the more insatiable the ear.
Mike-Power Myths

In your issue V-1, your microphone-powering data is pure myth!

First, microphone cables and other pro audio cables are never balanced when transformer coupled, they are floating. Connecting transformer center taps causes massive RF pickup problems. Balancing only works at low frequencies, so it is left to the 60-Hz engineers. Thus center taps are invariably left unconnected and the audio pair is left to float within its shield.

Phantom power is fed to and drawn from the cable through resistors. Look inside your AKG box for confirmation.

A DC supply of 9 volts won't even wake up most condenser microphones. 25 to 50 volts are required, depending on the microphone. I use five 9-volt batteries in series for a pair of KM-86s.

You wondered (page 8) why manufacturers of professional condenser microphones don't get listed in equipment directories? They do. Just look in Audio's current directory. It lists professional mikes from Neumann, AKG, Schoeps, Sennheiser and Sony.

P. H. Anton

You're right about Audio's microphone listings; we were wrong. Mea culpa. As for your other criticisms, you made it on some and struck out on some others.

First of all, when we referred to "balanced," we did not mean the specific case of a grounded-center-tapped circuit, but one where both sides of the line are identical and symmetrical with respect to ground. While many phantom-powering systems do introduce the power through identical resistors going to each side of the balanced line, others introduce it via a center tap on the preamp input transformer -- usually through a resistor, which probably serves to prevent the RF-pickup problem. And many microphones, including the AKG C-34, use the center tap of the microphone's output transformer to draw off the DC at the microphone.

The 9-volt figure was gleaned from AKG's own specifications for the C-34. Discuss it with them.

Grounding one side of a phantom-powered line is a no-no! Doing this not only destroys the symmetry of the line, eliminating its insensitivity to hum, but it also shorts out one side of the line, producing in that line a DC offset equal to the microphone supply voltage. If the microphone happens to use a transformer output (as does the AKG C-34), grounding one side of the line will cause the remaining side of the DC supply to drain to ground through the transformer secondary, causing probable core saturation and possible permanent magnetization of the core.

Recommended Components

I would definitely like to see you resume periodic publication of your Recommended Components list. To put it in every issue would be ridiculous, but it would be helpful to have such a list at least once a year.

Your Recommended Components listing can give a reader a good perspective on what you consider to be the better equipment available. I value your opinion quite highly -- probably moreso than that of any other "underground" magazine. Also, I've noticed that, the longer a piece of equip-
ment stays on your Recommended list, the more likely it is to become a "classic." I buy mainly used equipment, and get the feeling that if you can continue to recommend something for a while, I may be able to live with it for a while as well. As you well know, it is necessary to live with a piece of equipment for a while to learn its true personality.

W. R. Hitchens

Many years ago, selecting an audio system was likened to selecting a spouse. The analogy about learning its true personality by living with it is equally valid.

**LP Styli**

Does your record collection include any old LPs? If so, what cartridge do you use to play them? As you know, they were grooved for a 1-mil spherical stylus, but I'm not sure such a thing is available any more. Wouldn't a long Shibata or a 0.6-mil spherical bottom out? I've made out okay with a 0.7-mil spherical, but is that the best I can use?

Eugene Bershad

It probably is. We don't believe anyone in the US is making 1-mil styli, and are not convinced that there is a need for it. Even during the latter years of the mono LP, a number of cartridge manufacturers were finding that a 0.7-mil spherical provided cleaner tracing and lower surface noise than the 1-mil size that the groove was ostensibly tailored to. Bottoming was never a problem with the 0.7-mil stylus, but occurred increasingly often as the tip radius went below 0.5 mil.

If you are interested in using the best playback stylus you can get for the purpose though, you should know that a worn LP can often be "cleaned up" by using a small-radius stylus. This will contact the groove walls at a point below the wear line of the 1-mil stylus that were probably used to play it to death. Assuming the disc was never played on a badly worn (or chipped) stylus, there should be a line along the groove walls which is below the wear line and above the groove bottom. Selecting the right playback stylus will allow you to play that unworn part of the groove without contacting the bottom-of-the-groove curve. Bottoming causes a marked and sudden increase in surface noise and tracking distortion.

We have a special record-collector's issue coming up in the near future, and will include therein an address from which you can obtain custom styli of various sizes for any popular cartridge. That probably rules out Koetsus.

**Butterworth One**

One of the most noticeable trends in speaker design for the past few years has been the move towards 6-dB/octave crossover designs. I am bothered by this trend, because it begins to look as if we are seeing a replay of the mass-mentality that made "time-alignment" a household word not too long ago. What happens is that ONE aspect, often an important one but not the ONLY one, is singled out as the sine qua non of good performance, to the detriment of all other aspects. It's necessary to discuss 6-dB/octave design at length, to see whether or not this particular fad is really an important advance in design.

A 6-dB/octave rolloff is the simplest type of Butterworth (maximally-flat) crossover filter. The filter's output is down 3 dB at the nominal corner frequency, and rolls off at 6 dB/octave beyond that. This so-called first-order Butterworth response is the most gradual slope possible. (A
MUSICAL ACCURACY

S505 Mini Monitor

SPECIFICATIONS

FREQUENCY RESPONSE: 60Hz - 20 KHz±3dB
POWER: Min. 15 watts, Max. 200 watts continuous
SIZE: 10” x 6¾” x 6½”
PRICE: $250.00/pair (Continental U.S.A.)

If you have not yet heard the Orpheus S505, you are in for an amazing musical experience. The S505 expands the sonic limits to the outer reaches of the current state of the art, unlike any mini speaker on the market today. From the deep bass, that's right!, to the airy highs, the S505 remains smooth, accurate and musical. For the critical audiophile on a budget, the Orpheus S505 is a welcome answer indeed. Discover it today.

Orpheus

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second-order response rolls off at 12 dB/octave, third order at 18 dB/octave, and so on.) The appeal of the 6-dB/octave crossover is that the low-pass section of the filter is an exact electrical complement of the high-pass section. They add to produce flat frequency response, with NO phase shift. Another attractive aspect of first-order crossovers is that, if the drivers are purely resistive, the speaker will "look" like a purely resistive load to the amplifier, and any amplifier is "happier" looking at a resistive load than a reactive one. With a resistive load, we can expect more-efficient power transfer and less amplifier/speaker interaction.

With all these things going for them, why haven't first-order crossovers been used universally? Well, until recently, their problems have outweighed their advantages. This is primarily because first-order crossovers do not provide rapid attenuation of the driver's response outside the desired passband.

Take the woofer, for example. Above a certain frequency, which depends on the cone material and the over-all design, a woofer stops behaving like a piston. Different parts of the cone vibrate independently of one another and of the input signal, introducing a great deal of sonic junk. In other words, it is important to keep away from the woofer those frequencies it cannot cope with properly. With upper-range speakers, the problem is excessive diaphragm excursion. Frequencies below those which the driver is intended to handle will overload it, producing (again) excessive distortion. If we try to use a 6-dB/octave crossover, and choose too low a crossover frequency, we'll get distortion. In addition, the driver's mechanical crossover (the frequency at which its ability to move air starts to diminish) will get in the way, producing a suckout in the crossover region.

In other words, we could have serious problems. For each adjacent pair of drivers, we would like to roll off the lower one at as low a frequency as possible and the upper one at as high a frequency as possible. These requirements may conflict badly, and since first-order crossovers roll off so slowly, there will be a broad frequency range over which the drivers' outputs overlap. If their dispersion patterns are much different, or if they don't have similar sonic characteristics (colorations), there could be noticeable aberrations in the crossover regions.

Another problem, first pointed out by Siegfried Linkwitz, is that first-order (and all other odd-order) Butterworth filters tend to give the loudspeakers poor dispersion characteristics in the crossover region. In that region, the drivers are 90 degrees out of phase (because the 6-dB/octave treble rolloff and bass rolloff cause OPPOSITE 45-degree phase shifts at the 3-dB point). If the drivers are not time-aligned to correct for this, their combined radiation will direct the sound upwards instead of straight ahead, and the on-axis frequency response will not be flat. This poses little problem if the woofer/mid-range crossover is below about 250 Hz, since the drivers are (or can be) much closer together than one wavelength. But this doesn't hold at high frequencies; there may be problems with the upper-range drivers. The so-called Linkwitz-Riley class of quasi-Butterworth filters were developed to solve this problem. They give flat response and a uniform dispersion pattern, but at the expense of introducing phase shift.

As you can see, first-order crossovers work best with broad-dispersion wide-bandwidth high-
CC-3 STEREO/MONO AMPLIFIER

The CC-3 basic power amplifier is a new design, in the tradition of our highly acclaimed and successful CC-2 amplifier. Improvements in circuit topology, power supply current delivery and selection of component parts, provide a new standard of performance in an affordable, medium power audiophile amplifier.

These improvements, based upon years of experience in amplifier design and manufacture, enable the CC-3 to deliver effortless sonic performance. The CC-3 works effortlessly with low impedance loudspeakers including electrostatic designs which present a complex load.

At moderate levels, the CC-3 operates in the Pure Class A mode. As in the previous CC-2 design, the CC-3 utilizes negative feedback sparingly and also incorporates a feed-forward circuit for improved dynamic performance. Slew Rate Limit of 50 volts/microsecond in the stereo mode; 100 volts/microsecond in the mono bridged mode.

The circuit design features a dual-differential current sourced input stage with an intermediate and driver stage incorporating complementary transistors with a bandwidth of 100 megahertz. The devices in the driver stage are especially designed to match the transfer characteristics of the output transistors.

The original CC-2, because of it's excellent sonic performance and reliability, became a quiet favorite of professional users. There are hundreds of CC-2 amplifiers in use today in radio and television studios, discos, motion picture theaters and other professional applications. The new CC-3, with it's improved heatsinking and mechanical design, offers long-term reliability to meet the needs of audiophiles and professional users alike.

FUNCTIONS/FEATURES

Standard EIA 19" x 3 1/4 " rack mount
Generous heatsinking for thermal stability
Fused AC Mains Input and Loudspeaker Outputs
Front panel power switch
True Peak Clipping Indicators
Mono operation capability

275-A STEREO/MONO POWER AMPLIFIER

The 275-A amplifier is a lower cost version of the CC-3 amplifier. The 275-A is especially designed for the audiophile or professional user that may not require rack mounting capability or peak reading LED indicator circuitry.

The electrical performance specifications of the 275-A are identical to the CC-3.

SPECIFICATIONS

Frequency Response:
5Hz to 70kHz + 0, - 3dB

THD & IM Distortion:
Less than 0.1%; typically 0.05%, from 20Hz to 20kHz, both channels driven @ 8 ohms, at 70 watts per channel, FTC method.

Power Output:
70 watts per channel @ 8 ohms; 120 watts per channel @ 4 ohms; stable into 2 ohms (forced air cooling may be required under some use conditions with low impedance loads)

225 watts @ 8 ohms, 20Hz to 20kHz, with less than 0.2% THD & IM distortion in mono bridged mode. 250 watts @ 1kHz, mono bridged @ 8 ohms Stable into four ohm loads mono bridged

AUDIONICS of Oregon

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power-capacity drivers. As such drivers have become available, there has been a slow but steady swing towards first-order designs. The irony of it all is that one needs wide bandwidth so that one can then turn around and throw it away to accomodate the slow rolloffs of such crossovers.

First-order crossovers are NOT the panacea suggested by some manufacturers and reviewers. They require very-high-quality drivers to be successful, and for each problem they solve, they introduce another one. They may or may not be an optimum design; it all depends on the drivers chosen, and the way they are used.

I'll stick my neck out on this one: The sound quality of any speaker system depends PRIMARILY on the quality of the drivers used. A bad crossover can ruin the sound of good drivers, but no amount of tinkering with the crossover can ever make up for high-coloration, low-transparency, high-distortion drivers. There are probably many 6-dB/octave systems in which the improvements in transparency and coloration that could be achieved by a steeper crossover would more than outweigh any loss of quality caused by increased phase shift.

The moral of all this is that there is no single factor which conclusively determines a speaker system's quality. All but the most expensive designs represent many tradeoffs among conflicting design factors. Read the spec sheet if you like, but pay the closest attention to what you HEAR. That after all is what you're buying the speakers for in the first place.

William Sommerwerck

Test-Record Calibration

Your short article about checking test-record frequency response was interesting and useful, but it implied that you had invented the technique described. (Running the record at speeds

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Why Not?

If you support our views, why not support our magazine? Now that we're finally publishing monthly, and are starting to do the things we were only promising to do before, we expect you folks to show that you meant it when you said that, if we ever started publishing on schedule, you'd subscribe. Well, now we're publishing on time, which means you no longer have any excuse for not subscribing.

We've finally got our act together and don't insist that you cut up your magazine! Just fill your name in on the Business Reply envelope that comes in each magazine, enclose your check, and we'll make sure you get your next copy on time. If you're a first time subscriber and don't want to begin with Issue V-1, be sure and let us know.

35
The Sony ESPRIT TA-E900 Stereo Preamplifier

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other than normal.) I call your attention to a paper by Bernard Jacobs of Shure Bros., in the June 1970 "Journal of the Audio Engineering Society," which described this procedure in some detail. A Xerox of the paper is enclosed.

I think you should give credit where due. Arthur Clement

We did not mean to give the impression that that clever little trick was our idea. We were aware of Mr. Jacobs' paper, but are also aware that Paul Weathers (of the now-defunct Weathers Industries) was using the same technique as early as 1961, and said that HE had learned it from someone else. So while we would be happy to give due credit for it, who do we give it to?

Incidentally, SIR, we notice that your name is not among those of our active subscribers, and assume that it is your intent to subscribe at the earliest possible opportunity. That IS your intent, is it not?

Decent Covers

How about some decent-looking covers for the Stereophile. I've had it with the present appearance and would like to see a return to yesteryear.

David J. Thompson

We agree, but we cannot, single-handedly, come up with enough interesting covers to adorn 10 issues per year. So, we are asking all of you out there to keep an eye open for photos that we could use. We'll pay $5 for each one we use.

Shure Styli

Will the improved stylus of the new V-15-V fit my V-15-IV? I hate to pay the full price for the new cartridge if I can get its perfor-
mance for the price of a stylus alone.

Charles Steele

Do you really think Shure Brothers would still be around if they were stupid? Not only will the V-15-V stylus not fit your IV (without major surgery), it wouldn't give you the benefits of the V anyway, because part of the reason for the V's improved high-frequency response is its use of lower-inductance sensing coils. You'll just have to buy the V or put up with the IV.

The Audio Arnold

Were you people aware, when you made a rather rudely flippant remark about an Arnold Schwartz in your "Audio Chronology" (Issue V-3), that there is indeed such a person in the audio field, and that he is president of Micro-acoustics Corp.? Did he (or his wife) actually have what you said he had, or was that merely a tasteless put-on? R. Williams

JGH, who wrote that piece, is positively radiant with embarrassment. That character was supposed to be fictitious, as was the event ascribed to him. We wanted a Jewish name, because there are more Jews in audio than Arabs in oil.* But we thought we had picked a name that was not a real member of the audio community. We, and specifically JGH, apologize profusely to the real Arnold Schwartz, and promise in future to stick with Smiths, Joneses and Does.

*This is probably because a disproportionate number of Jews have the kind of abiding love for good music that others profess to while choosing to listen to pap, crap or dross.
Audio Puzzler

Jim W., recently bitten by the audio bug but a little strapped for funding, decided to assemble his own loudspeakers. He built a pair of bass-reflex enclosures from plans in a hobbyist magazine, and carefully tuned and damped the ports as instructed.

For the upper ranges he chose horns, partly because he liked that kind of forward sound and partly because he couldn't afford a high-powered amplifier, so he needed a high-efficiency loudspeaker. The tweeters were fairly large units which would allow him to cross over at 800 Hz.

He fastened the horns atop the woofer enclosures, and then fired up the system in preparation for the last step: Woofer/tweeter phasing.

He had read that woofer/tweeter misphasing would cause a large phase-cancellation dip at the crossover, making the sound seem to recede from the listener. Yet when he tried, first one tweeter polarity and then the other, he was nonplussed to hear no difference at all!

"Maybe," he thought, "it is just that I haven't learned what to listen for. I need an A/B comparison." So saying, he lashed up a phase-reverse switch near one end of a long wire, and ran that to one speaker so he could switch the phasing from his listening seat. STILL no difference. Both positions of the polarity switch still sounded the same.

Stymied, he telephoned an audiophile friend and explained the situation. "Oh, that's an easy one," his friend said. "I can tell you right off what's the matter."

What WAS the matter? The answer will appear next month, but until it appears, we'll give a free subscription (or renewal) to the first reader (present subscriber or not) who writes us with the correct answer.

GIVE US A HAND

We're working hard to get out your issues of Stereophile on time (with not universal success, as you realized with issue V-4). We also have in mind great plans for improving the appearance of the magazine, expanding the range of products we review, and even paying ourselves! All these things take money, and raising subscription prices isn't the answer. Even advertising can only go so far in a magazine of limited size -- you wouldn't want to see your 40 or 44 page Stereophile filled with 15 to 20 pages of ads, would you? The answer is a much larger group of subscribers, and we're counting on our present subscribers to find those new ones for us. Of course, we're not counting only on you (you may have seen our ad in Fanfare or the one upcoming in The Absolute Sound), but you're our greatest (and almost only) resource. Now that we're providing Business Reply envelopes in each magazine to make subscribing easy (you don't have to sacrifice your precious magazine to a friend's scissors), why don't you take one of those envelopes and give it to a fellow audiophile so he or she can subscribe -- of course, a little sales pitch won't hurt either! To top it all off, we're even willing to pay you: not in cash, but in kind. Borrowing a page from another's book (magazine), we'll happily extend your subscription by two (2) issues for every new subscriber who puts your name and address on their subscription order (along with theirs of course). Just think: all you need to do is find a good home for those ten business reply envelopes you have coming, and you may never need to renew your subscription! And then, while basking in your new found wealth-to-be, drop us a line as to what direction you'd most like us to take in the future:
glossy paper, color covers, reviews of fancier equipment, reviews of more modest equipment, reviews of your favorite 15-year old speaker, whatever you want. Then we'll know what to do with all that wherewithal you generate for us!

(con't from page 5)
manufacturer called Exposure. Add that name to Focus, Aparate, and (Sonic) Developments and the audio field begins to sound like a photographic industry.
That's all for now. More next month.

JGH

CLASSIFIED ADS
You Stereophiles out there who aren't using Audio Mart are missing the boat when it comes to getting an ad published quickly and cheaply. For instance, I received the following ad just today (July 12th) and it's going to make it in the issue being mailed July 30th!


Audio Mart

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dbx 118 DYNAMIC RANGE COMPRESSER/EXPANDER; very good condition, $125. dbx 3BX-R remote control for 3BX expander; brand-new, never used, $125. Signet MK12T moving-coil transformer; 3 step-up ratios plus by-pass, good condition, $125. Pioneer RT-2022/44 semi-pro open reel deck; records and plays both half-track stereo and quarter-track quad; interchangeable head blocks; full range of bias, EQ, and level controls; self-synch; 7½, 15 ips; remote control included; many other features, carefully maintained, will adjust for buyer, $1000. Freight included for all products except Pioneer; personal delivery of Pioneer included in price if buyer lives within 200 miles of DC. William Sommerwerck, c/o Stereophile, PO Box 1948, Santa Fe, NM, 87501.

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