AUDIO BASICS

The Complete 1983 Back Issue Set.

VOLUME TWO NUMBER ONE
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This issue begins the second year of *Audio Basics* and again I promise you 12 monthly issues of information to make your audio system work better. Thanks for your support, comments, and suggestions. Over 200 of you have renewed so far. Among topics to be covered this coming year will be a do-it-yourself project for the Dyna PAS-2, PAS-3, PAS 3x series preamplifier (our SUPER-PAS preamplifier), a discussion of digital audio and what its technical limitations are (reliability may be the main problem), things you can do to make your listening room work better, and (if enough of you express interest) we could spend an issue on "home computers" going through what they can do, more importantly what they cannot do, what pitfalls to avoid, etc. I would need to hear from many of you before spending an issue on computers.

Lets start this year with a few follow-ups that may be of interest. First of all, the Longhorn stabilizer bar for your phono cartridge we described in Volume One, Number Four can be improved if you have enough room in your setup for a longer bar.

Essentially, the article suggested making a 1.5 inch stabilizer bar. Many of you have done this and the reports back to us have been very favorable. We have been playing with the bar length, and in essence, the longer the bar, the better the sound quality. Those things the stabilizer bar improves keep improving – dynamic range, imaging, tracking ability, tonal balance. I suggest you try making a longer bar. The main constraints are that it not overhang the record when the arm is on the rest so it doesn't become a record scratcher, that it not hit the spindle at the end of the record, and that it not get so heavy that you cannot balance the tonearm. I am using a 2.75 inch long bar on my own Grado here now. Obviously a longer Longhorn bar isn't possible in our production cartridges; packaging for shipment becomes impossible and we must be conservative in allowing operational room on the many kinds of turntables out there. However, if you have the clearance room on your turntable, we urge you to try making your stabilizer bar longer, the results may surprise you.

By the way, for those of you that have been dubious of the value of the Longhorn bar, I could note that we sold over 500 Longhorn Grado cartridges in 1982 with our 30 day satisfaction guarantee or your money back offer, and the total returns for a refund were three! I suggest this is reasonable evidence that the stabilizer bar works.

Next, briefly, a follow-up on Wonder Caps. We have finally had the opportunity to listen to Wonder Caps. We must apologize, for we are forced to admit that we could easily hear the sound of the two 10 µF Wonder Caps a client gave us. That is the good news. The bad news is that what we could hear was that these samples of Wonder Caps were very microphonic! Tapping on them makes one of the neatest "bong - bong" sounds I have heard since Telefunken quit making 12AX7 vacuum tubes. With a preamp full of Wonder Caps, you may hear your cat walking across the carpet. Did I like the changed sound? Nope!

I have had a bit of negative feedback on audio system troubleshooting. It seems like some of you consider the subject boring. Maybe, but we just got in a Transcendence Preamplifier to "repair" in which the problem was that "nothing worked" except for the selector switch. The volume control, balance control, filters and tone controls all were "dead" and the output was at a constant level. The "problem" was, of course that the user had hooked his amplifier to the tape outputs of the unit (which are ahead of all the mentioned functions) rather than to the audio outputs. I wish he had read the troubleshooting issues, would have saved a round trip of shipping of the unit and the checkout fee. Anyway, we will give up on troubleshooting with this issue, after going through the second troubleshooting problem.

There were only two correct answers to the second troubleshooting problem, in addition to Mr. Hugus' original answer. They were from John Moyer of Colorado and from Mitchell Halperin of Massachusetts. The following is an edited version of Mr. Halperin's response:

"AUDIOKLUTZ SOLUTION. The audioklutz spent the evening before listening to space between the instruments on the Telarc 1812 recording, and at the end of the evening while in a happy stupor, tripped over his 0000 gauge speaker wires, breaking the connection at his left speaker, and at the same time jarring the amplifier enough so the left input cable became ungrounded (partially pulled out). When he turned on the system the next evening, there was no sound from the left speaker because the speaker cable was disconnected at the left speaker. When he switched speaker wires at the amplifier then the right speaker makes a large hum (because of the open input ground) and then its speaker fuse blows. There is no sound from the left speaker as the klutz still has not noticed the disconnected speaker wire at the speaker. No equipment is damaged thanks to the fuse. When the audioklutz brings in a friend's amplifier and hooks it up in his system, there is no sound because the fuse is blown on his right speaker, and the connection is still broken at his left speaker (remember in troubleshooting, first things last!). Friend's amplifier is then transported back to friend's system, and in the haste to reinstall, the very stiff magic interconnect cables between amp and preamp are stretched a bit too far, so that an open ground occurs at both amplifier inputs (don't laugh, I personally saw this very thing happen with a famous reviewer hooking up one of our amps in his own home system using magic interconnect cables, FVA). Of course now a loud hum comes from both of friend's speakers (open input grounds) until both speaker fuses blow and silence prevails. When the audioklutz brings his speakers to a second friend's system only the left speaker works as it had been protected from the earlier abuse by an open speaker wire connection. The right speaker is dead as its fuse was previously blown. Inasmuch as the built in protection equipment worked in spite of the attempts at abuse, the only problems are blown fuses in the original right speaker, and in both of 1st friend's speakers. Moral: always check connections and fuses as first step in troubleshooting."

Both Mr. Moyer and Mr. Halperin will get a two issue extension to their *Audio Basics* subscription. With this we will end troubleshooting for the time being. However, dear reader, if you do have a problem with your system you cannot resolve, you are welcome to call us, we will try to help you resolve equipment problems. There is no charge for our telephone time. Note however that I cannot take the time to answer long detailed individual letters, my typewriter is a timesink. Please call if you need help or more detailed advice.

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The main topic for this month is HOW TO EVALUATE LOUDSPEAKER SYSTEMS.

With hundreds of brands of loudspeakers available, with hundreds of conflicting advertising claims made, with conflicting reviews of many in both commercial and underground magazines, together with the difficulty of separating loudspeaker performance from that of the amplifier driving it, compounded by the difficulty of distinguishing the speaker's traits from that of the room it is in, we suspect that finding a "good" speaker system can be a bit confusing. The following is a set of constraints and observations that we hope will help you sort out the claims and find a loudspeaker that meets your requirements.

Constraint #1. THERE IS N0 SUCH THING AS A PERFECT LOUDSPEAKER SYSTEM. Even after sorting out the 90% or so of all speaker systems that are obviously "Grot" (hopelessly non-linear and/or colored) the remaining 10% all have obvious imperfections. It is just that their imperfections are less obvious and annoying. Thus, after you have discovered a reasonably competent system at for example, $1000 a pair, be very sure that the advantages of a more exotic system (perhaps priced at $2000 a pair) are really advantages, and not just a minor rearrangement of nonlinearities. Do not neurose yourself into paying too much for new speaker systems based on the opinion of others that those exotic speakers are "better." They probably are not better, just different.

Constraint #2. ARE YOU SURE YOU REALLY NEED NEW LOUDSPEAKERS AT ALL? Are you certain that the annoyances you now hear from your speakers are the speaker's fault? There are many very competent older speaker systems around that are difficult to drive and thus may not please you because the equipment driving the speakers isn't linear enough. (Remember, garbage in = garbage out). Many of the older AR speakers (AR3, AR4-AX, AR2A, etc.), many Bozak models, KLH-6, Dyna A-25, 35, 50 series, and similar were neutral speakers with good definition IF driven and located properly. Some speakers, such as Dahlquist DQ-10s, can be made so good with proper panel damping (see *Audio Basics* Volume One, Number One) that it becomes difficult to find anything that is really "better." The most interesting characteristic about a neutral speaker system is that it stays good for a long, long time and tends to become better, the better the drive equipment becomes.

Because of Constraints #1 and #2, we tend to evaluate speaker systems under two different set of considerations. Consideration #1 occurs if you already own a competent speaker system (perhaps in the top 10% of all available). Then the cost to you to actually acquire a different speaker that is significantly better, and not just a bit better or "different" may be quite high. We will suggest to you that you re-evaluate your premises and perhaps concentrate on driving the speaker you now own with greater linearity. It is very probable that a higher quality phono cartridge or power amplifier may do more for you than a new speaker system. Consideration #2 occurs when you do not now own a competent speaker system, then we will suggest you aim for a higher standard and select from perhaps the top 2% if that can be done at no significant increase in cost. Thus, if you already own, for example, a Dahlquist DQ-10, we probably would not suggest that you get new speakers, however, if you are shopping for new speakers, we probably would suggest that you can do better than the DQ-10 for the money. Got that straight?

Constraint #3. WHEN YOU PURCHASE NEW SPEAKER SYSTEMS, BUY THE VERY BEST YOU CAN AFFORD, AND BUY THEM T0 KEEP THEM LONG TERM. The old adage that nobody ever later regretted buying quality is true with speaker systems. There are many people happily using Quad ESLs, Bozak Concert Grands, and other fine speakers designed 20 years ago, that have not had to spend a cent on loudspeakers for years. Their speaker system cost over the years has been very low, even though they bought an expensive product in the first place. If, for example, you are considering B&W DM 12s ($600/pr.) vs. B&W DM 14s ($900/pr.), – both have nearly identical definition, mid-range and highs but the DM 14 plays somewhat bigger, louder, and with greater clean bass extension – then go for the DM 14 if you can afford it. The $300 difference won't mean much over 10 years, the deeper clean bass will always be meaningful.

Constraint #4. DO NOT NEUROSE IF YOU CANNOT AFFORD AN EXPENSIVE SPEAKER. D0 THE BEST YOU CAN WITH THE FUNDS YOU HAVE AVAILABLE. Although this may seem in conflict with Constraint #3, it is not. There is always a more expensive speaker, you have to stop somewhere. We have many clients who have carefully selected a relatively inexpensive system (a Technics SL-B202 with a NAD 3020 and a set of B&W DM 10 speakers, total system cost about $600 for example), and have quality good enough that the system vanishes and they listen to music, not equipment. They do not really care that another $1000 could have bought deeper bass and greater dynamic range, they enjoy what they have, because it is worth enjoying. Most of us can never afford a top of the line Mercedes-Benz automobile, this doesn't prevent us from shopping for the best car we can afford, and enjoying what we acquire.

Constraint #5. AFTER YOU DO PURCHASE NEW SPEAKERS, AVOID HI-FI SHOPS! THERE WILL ALWAYS BE A BETTER SPEAKER. T0 ENJOY WHAT YOU HAVE LONG TERM, AVOID EXPOSURE TO HI-FI SALESMEN AND THE LATEST NEW SPEAKERS. Don't go listening to other speaker systems until you have independently decided that what you have isn't good enough for you.

VAN ALSTINE'S OBSERVATION 0N HOW YOU CAN TELL IF YOUR (OR A FRIEND'S) HI-FI SYSTEM IS "GOOD ENOUGH."

When you visit your friend, is the hi-fi system turned on? Is your system turned on? If so, it is probably good enough. If it is off, it probably isn't.

Minor observation, if your budget is really tight, don't overlook Radio Shack. Some of their very inexpensive speakers (under $100/pr.) are quite respectable. The main difference between Radio Shack and the major discount hi-fi chains is that the major "mid-fi" chain's price their speakers at "were $500 a pair and now on sale for $200 a pair." At Radio Shack the equal quality is $89 a pair. Did you know that Radio Shack's are the largest selling brand of speakers in the U.S.A.?

Constraint #6. IF YOU ARE SHOPPING FOR A COMPLETE NEW SYSTEM, SELECT THE SPEAKER SYSTEM FIRST. It is very discouraging to find, after you have purchased the latest and greatest exotic power amplifier, that the speaker systems you subsequently fall in love with require more power or greater stability than your amplifier delivers. Conversely, if you are happy with your present amplifier and have no desire to replace it, then bring it with you in shopping for speakers, and use it in making all subjective evaluations. Remember, the loudspeaker "quality" cannot readily be separated from the amplifier driving it. Remember too that a bad amplifier will make a good speaker sound bad. By bringing along your own amplifier, you will be assured that the speaker you choose will work well with your amplifier, and you will avoid overlooking (assuming that your amplifier is of good quality) those good speaker systems in your market area that sound "bad" in the shop because the electronics in that shop are not so hot.

Two comments regarding the above constraint: We have problems with inexperienced shoppers in our local area because there is another shop that handles B&W speakers but demonstrates them on "grot" mid-fi receivers. Often, we get calls asking what speakers we handle. When we mention B&W, too often the response is, "I heard them at ——, they sound harsh and boomy." Our efforts to suggest that perhaps the person should listen to them again here often is futile, once a person has heard a good speaker sound bad because it was driven by bad electronics, it is hard to change his mind. Thus, many mid-fi shops have best success selling muddy, low definition speakers, because they tend to mask the harshness of their scuzzy receivers. The second observation is that if you do take your own amp or receiver along to evaluate speakers, and if several speaker systems that have a good reputation do seem to be harsh, muddy, or boomy with your amplifier, it may be time to suspect your amplifier isn't too wonderful. In general, electrostatics are not harsh or "steely," KEFs and B&Ws do not have boomy bass, and Dahlquist DQ-10s will play clean deep bass. If this is contrary to your previous experience, suspect your amplifier.

If your present amplifier is good, we suspect you will finally locate several different speaker systems that are acceptable to you, and that sound very similar to each other. Remember, if several different speakers were perfect, they would sound identical, thus it isn't surprising that those speakers that approach true linearity are quite similar in overall characteristics (hopefully, they have no "characteristics"). All other things being nearly equal, pick the lowest cost speaker meeting your constraints from the dealer best able to provide good long term service. Caution, if one of your choices sounds distinctly "different" from several others, it is probably wrong!

Now lets go through a series of "go/no-go" constraints regarding speaker systems, first dealing with the physical construction quality of the unit in question. We will take the easy to identify "pass/fail" tests first:

Test #1. If the system in question has many more individual speakers than necessary (a 12" woofer, a 10" mid-bass, two 6" mid-range, three 3" tweeters, and a 1" "super-tweeter," for example) you can be certain the system is bung, and is designed to sell to those who purchase hi- fi by looking at the equipment, rather than listening to it. The sales theory is "the more speakers in the cabinet the better." Actually, it is seldom necessary to use more than three drivers (woofer, mid- range, and tweeter) to build a competent full range system. In addition, the greater the number of drivers, the more difficult it becomes to design a good crossover. Actually, if you are shopping for a low priced speaker, you are generally better off sticking with a two-way design (simple woofer and tweeter) as the design of a good crossover for an inexpensive two-way system is much easier to do than in a multiple driver system. We find that in many inexpensive three-way systems, the mid-range sounds like it belongs in another speaker system entirely, and hopefully far away from you. Two specific exceptions – the Dahlquist DQ-10 successfully integrates a multi-driver system and some larger Infinity speakers use multiple small planar panels successfully to achieve adequate acoustic output.

Test #2. Have the salesman pull the grill from the system so you can examine the construction quality carefully. If the grill cannot be removed, reject the system as this indicates two major potential problems. 1) If the speaker needs service, it may be necessary to return the whole thing to the factory. This can be expensive, slow, and remember that truck freight shipping damage is not uncommon. 2) The speaker manufacturer may be trying to "hide" his poor quality or very inexpensive drivers and construction techniques from you. Remember, he cannot hide them from his competition – all another company has to do is buy a set and saw them apart. He can only hide grot from the end user. We know of one pretty good two way column system built with a compressed paper cabinet and very inexpensive imported drivers, in which the whole works is "hidden" by a foam, wrap-around grillcloth. The overall design is clever, but this $800+/ pair system should sell for half that, and would, if the end user could tell how inexpensive the construction of the system really was. We consider that the price of a given speaker system must be justified by excellence in three separate areas: construction quality and workmanship (including ease of service), technical excellence, and sonic excellence. We consider a system overpriced if it falls down in construction quality and its asking price is only rationalized by its satisfactory sonic quality.

If the grill can be removed, examine the unit very carefully. Look first at the drivers themselves and the way they are mounted. The drivers should be mounted on the front side, or flush with the front panel. If they are mounted from the inside of the panel, with the mounting hole in front of and surrounding the driver, there will be unnecessary refractions and colorations from the panel edges – the speaker is actually mounted in a short "tunnel" and it will go honk - honk. That kind of construction is disqualifying. Next look at the mounting hardware and methods. A quality product will have drivers mounted with large machine bolts into molded or pressed in T-nuts on the mounting panel. The frame will be gasketed between the speaker and the panel. In quality construction, it will be possible to replace individual drivers in the future without damaging or marring the cabinet. Avoid those speakers put together with hot melt glue and wood screws as future service requires prying things out, and wood screws into particle board don't hold a second time.

Next look at the dust cap (small dome over the voice coil in the middle of the woofer and/or mid-range). Tap very gently on it. It should be soft and inert. If the dust cap is "hard" it will be a significant source of distortion and indicates undesirable construction. Examine the speaker cone material. In general the most neutral speakers use synthetic materials such as Bextrene (soft black plastic), polypropylene (soft clear or translucent plastic) or fibre materials that are obviously damped and/or coated. It is generally thought that the synthetic cone materials are more opaque to sound and better keep the internal box reflections from coming back out into the room, thus reducing the honk or "box" sound. Few quality speakers are built with hard paper cones. Regarding polypropylene cones, inasmuch as this material is very difficult to glue, we suggest you stick with products from a source you are sure will be there a few years later, as there isn't enough field experience yet to be sure that polypropylene drivers will hold together long term. Avoid speakers with "foam" edge surround materials (the flexible ring connecting the cone to the speaker frame). Our experience is that these materials deteriorate too quickly, especially in inexpensive speaker systems.

Now examine the metal material the frame of the speaker is made from. All other things being equal, a cast frame is stiffer and freer from resonances than a stamped framework and can indicate that the manufacturer cares a bit more about his product. If the speaker is designed for easy service, the salesman may even be willing to remove a driver from the cabinet so you can see how the other side of the driver is made. Again solid, stiff, and inert structure is important. We would like to see massive magnets (the expensive part of the speaker) and good sized voice coils. Some speakers feature 12" woofers that look great from the front, but from their rear side one finds flimsy sheetmetal framework and a magnet that is a refugee from a car radio. The quality of the parts you cannot normally see should be as good as those you can see, if not, the manufacturer is trying to fool you. Look into the cabinet while the driver is out. You should see solidly braced construction (in the best speakers there will be heavy internal cross-braces and heavily damped panels). Avoid speakers damped with cheap fiberglass insulation (the stuff you have in your attic), it just doesn't work as well as acoustic foam, wool, or polyester "angel's hair" and is an indication of corners cut that should not have been.

If you can spot the crossover network inside the speaker, pay attention to it too. Most quality speakers will use circuit boards designed to accommodate their crossover parts. This insures consistency and reliability of the product, and again is a clue of a little extra effort on the part of the manufacturer to produce a good product. If the crossover parts are randomly tacked to the back panel with hot melt glue and foam backed tape, if crossover capacitors are built up by wiring a bunch of polarized capacitors back to back and in parallel into an ugly glot, avoid the product. You will be able to do as well with a different product that has better construction quality for the money. One exotic speaker manufacturer we know built crossovers using old car starter solenoids for his coils and what appeared to be what was on sale that week at the local electronic surplus store for capacitors, all held together with hot melt glue and black tape. There was some controversy over the quality of his $4000/pr. systems. I suspect it was because it was difficult to get any two samples the same using this construction method.

Do not neurose too much over the kind of internal wiring in the system. The wiring should be tidy, color coded for reduction of assembly errors, and with solid connections to the crossover and drivers. “Magic cable” is no more important inside the speaker than outside, and magic wiring cannot make up for bad design and workmanship, and a claim that the speaker is wired with some kind of exotic cables may just be a ploy to make you overlook the inferior characteristics of that product.

Next take a close look at the tweeter but don’t touch, they are fragile! In general, the best tweeters seem to be small “dome” type (little paper cone tweeters usually don’t hack it). Tweeters located in plastic horn throats, or with plastic diffusion fins or dingleberries in front of them usually create more spurious resonances than music. An exception is the tweeter in the outstanding Celestion SL-6 speaker system, the metal dome tweeter is protected by a front slotted plastic grill and it still works just great. Keep in mind please that all of the above suggestions are not “etched in granite,” it is possible to make a good system violating all of the above constraints, it just isn’t very likely.

Test #3. Does the speaker system protect itself from abuse and accidents? At a very minimum, we would like to see built in fuses in the speaker so that accidents, “parties” and the like blow fuses instead of woofers and tweeters. The more sophisticated speakers now have internal electronic protection circuits that sense input conditions and shut off the speakers under drive conditions that would be damaging, making the speakers nearly “idiot-proof.” How about the input connectors on the system? We would like to see heavy dual banana jacks or five way binding jacks in a recessed panel so there is nothing to accidently break off. Dinky screw terminals and spring loaded clips belong in the mid-fi shops, not in your home.

Consider the overall construction quality. If it looks like a reject from high school wood shop (some very expensive speakers do) you should reject it too. It appears that B&W sets the standard for high quality cabinet construction and finish. You should examine a B&W speaker if for no other reason than to learn how well a system can be built. Do not reject a vinyl finish if cost is a major consideration. A carefully done vinyl can look and be built better than a half-ass job with real wood veneers. Remember, all speakers are particle board underneath the finish layer. Solid furniture grade woods are not suitable for the structure of the cabinet. All other things being equal, the heavier speaker of the same size may be better and the closer the cabinet sounds to being a solid block of cast iron when you rap on its top and sides, the better. You cannot make the cabinet too stiff and inert, this is why the B&W 801F midrange cabinet is made of concrete!

Something else to consider is if there are high quality stands available for the speakers. Both Celestion and B&W along with a few others, make their own speaker stands, designed to safely support the speaker off the floor and away from surfaces, and to optimize the sonic performance of the speaker. Others have stands that are a shaky afterthought. Guess what you should choose?

Finally, is the speaker you are going to buy the same as what the manufacturer advertises and supplies to reviewers? We know of one company who’s expensive system as displayed at trade shows, given to reviewers, and photographed in brochures and advertisements is very different than the product shipped to dealers and sold to you! The “display and reviewer’s” version has double thick cabinet walls, and the brochures picture this cabinet, with a “picture frame” cutout on the outer front panel for the drivers. The speaker they sell you has a much cheaper single thickness cabinet, a flat front panel, and much poorer sound quality. The most polite term I can think of is FRAUD! Compare the brochure to the speaker, is it exactly the same?

Well! Another issue of *Audio Basics* through the old IBM 75 [1990 note: scanned with a H-P ScanJet Plus scanner into a Macintosh IIfx, translated into text with OmniPage 2.0, edited and spell checked in Word 4.0, and finally imported into PageMaker 4.0 and formatted and re-edited again for this concatenated annual back issue set] and we have barely scratched the surface of our speakers. (By the way, you can hide that scratch by rubbing it gently with a chunk of walnut - the real nut you know, the kind you eat). More next month.

Frank Van Alstine

VOLUME TWO NUMBER TWO FEBRUARY, 1983

Hi again, we will continue with loudspeaker systems this month, concentrating on the engineering and design aspects. But first a little “equipment review.” The purpose of *Audio Basics* is not to bring you “subjective” reviews of audio equipment, but occasionally we will point out products that have unusually competent engineering (such as the Celestion SL-6), and this month a product that has, in our opinion, unusually incompetent engineering.

This past month, one of our clients brought in for us to evaluate a preamp that has twice recently gone into full bore oscillations destroying his power amplifier. The preamp is an RGR preamplifier. One highly regarded by the underground golden ear press, you know, those that evaluate by listening only, know nothing about engineering, and never take the covers off the equipment to even look at the quality of the equipment they claim to be the latest and greatest. We did take off the cover and bottom plate to make an engineering evaluation of the RGR unit. I haven’t seen such a mess since I quit repairing K-Mart walkie-talkies. We made a count of engineering “kludges” in this so called “state-of-the-art” preamplifier. There were:

> 110 parts tack soldered and flying in mid-air in other than their planned original circuit card locations.
>
> 30 ferrite beads added to input, output, and interconnect wires to attempt to suppress oscillations.
>
> Two unsoldered connections to the main output grounds (this after the unit had been back to RGR for repairs).
>
> Several foil cuts on the main mother board where original foil layout had been replaced with jumper wires.
>
> Output relay that was defective as installed (contact missing) and “repaired” by bending down the contact backing plate to make a crude connection.
>
> Six audio cards installed on the mother board in oxidized edge connectors with no provisions for mechanical location and integrity other than the edge connector contacts.
>
> A DC protection circuit that could be defeated by plugging the unit in with the power switch previously turned on – causing DC offset to the rails in spite of the output relay. Don’t use this unit on a switched AC outlet to turn it on or off.
>
> Substantial transient overshoot and ringing when operated into a standard IHF load (10,000 ohms in parallel with 1000 pF.).

We cannot understand these construction methods. The six active circuit cards are so small (about 2" x 2") that the tooling cost for a complete new layout including redoing the PC card artwork, photos, drill templates, etc., would be less than $500.00. The cards, in quantity (single sided epoxy fiberglass) are less than $3.00 each. Thus the cost to RGR to completely retool to put the parts reliably on the PC cards and get repeatable and stable results would be less than the labor cost to tediously hand wire even two or three of the units with 100+ parts flying in mid-air. Not only is the unit infeasible from an engineering layout standpoint, but it is very uneconomical to produce and must present terrible quality control problems. There is no valid reason for this kind of construction. Yet the golden ears never bother to look inside and claim this is a "wonderful" piece. Even if one reviewer's sample should happen to be "just wonderful sounding," we doubt if any two units can come out the same. Could this be part of the reason there is so much debate and arm waving among various audiophiles as to what equipment "sounds good"? Is the six blind men and the elephant syndrome at work?

O.K. readers, I wonder if some of you are getting tired of hearing only me yak away. I am considering giving some of you a forum in *Audio Basics*. Here is what I want: Articles of general interest to our readers such as construction or modification tips that work (but only with objective information as to why they work) – no subjective "arm-waving" about something that is "just wonderful" – this eliminates subjects such as magic wires, cables, and capacitors. Articles calling attention to new products showing interesting engineering qualities. If any of you now have a digital tape deck or converter (such as the Sony PCM-1, etc.) or a digital disc player we would like to hear of your opinions of it – in particular about its reliability, how it handles dropouts, the reliability and availability of software, etc. I would like to report on equipment service and be able to inform readers of those service agencies you have had fine results with, and those that have given you problems (if possible, we will try and help with the problems). If some of you disagree with a report of mine, let me know. I will print your rebuttal, but only if you include objective observations. "It sounds wonderful – it doesn't sound wonderful" cuts no ice here unless you have objective explanations as to why. Have any of you had good (or bad) results with loudspeaker kits? Tell us about it. How about "modifications"? Have any of you got something that is better (not just different)? Send us the schematic and let us know why it is better. Let's hear from you.

Now, on to LOUDSPEAKER SYSTEM EVALUATION (continued).

Let's see now, we can choose horn speakers, planar speakers, omni speakers, dome tweeters, cone tweeters, ribbon tweeters, full range electrostatics, hybrid electrostatics, woofers facing down, up, sideways, back, reflecting speakers, free standing speakers, wood cabinets, concrete cabinets, lead cabinets, composition cabinets, big speakers, little speakers, bi-amped speakers, tri-amped speakers, kit speakers, imported speakers, ionic speakers, electromagnetic speakers, round speakers, pentagon shaped speakers, big sub-woofers, little sub-woofers, satellite systems, full range systems, ported speakers, transmission line speakers, acoustic suspension speakers, infinite baffle speakers, self-powered speakers, vinyl speakers, equalized speakers, laser designed speakers, 2-way, 3-way, 4-way, and 5-way speakers, dual voice coil woofers, flush wall mount speakers, corner speakers, mini-speakers, refrigerator sized speakers, running a gauntlet from $39.00 a pair to $3900.00 a pair and more. Confusing isn't it?

We better start with a basic evaluation of some common speaker designs saving the most common speaker in a box designs for last.

PLANAR LOUDSPEAKER DESIGNS. Those systems using flat membrane drivers (usually "free air" mounted – no cabinet enclosure) with acoustic output from (usually) both the front and rear sides of the membrane. The most common type of planar system is:

ELECTROSTATIC LOUDSPEAKER. Typically a very large free standing panel (Quad ESL and Acoustat are type examples). There are no conventional woofers, tweeters, magnets, voice-coils or suspensions. The system operates by having a very large thin plastic film stretched in a supporting framework. The film is coated with an electrically conductive material. On each side of the film diaphragm is a series of electrically conductive grids. The speaker system has an active high voltage DC power supply (connection to line AC is required). The power supply puts out a several thousand volt DC potential which is fed to the diaphragm and/or grids so that one grid is positive in relation to the diaphragm which is positive in relation to the other grid. The audio signal modulates the relationship of the static voltage potential between the diaphragm and grids which causes the film diaphragm to be attracted towards one panel and pushed away from the other panel, the physical movement of the diaphragm causing acoustical output.

Design advantages: The film panel is very low mass and usually very well controlled from mid-frequencies up, as all points (except at the edges) are directly driven. In comparison a conventional cone – voice coil – magnet speaker is driven at the center only. Thus there is no conventional "cone breakup" that is possible with a conventional speaker. The typically very large diaphragm (usually the size of a door) presents a pleasing (to many) "large" sound stage. Inasmuch as there is no conventional speaker box cabinet structure, the unwanted "box" resonances of a poorly constructed conventional speaker system are absent. Some people think the acoustic output from the back side of the panel, reflected from the walls behind the system, enhances their perception of imaging and depth. Electrostatic speakers have a general reputation for high quality, high frequency performance and some small electrostatic speakers are used for tweeters in hybrid systems (Janzen, RTR are examples).

Design disadvantages. Because a very large voltage swing is necessary for adequate acoustic output (+ and - several thousand volts) the electrostatic speaker cannot be directly connected to an amplifier. Consider that a 200 watt amplifier has just a + and - 60 volt output swing. Some have tried direct connection to a high voltage vacuum tube amplifier but without success as a several thousand volt per microsecond slew rate is also required, and the high voltage vacuum tube amplifier-rarely can exceed 10 volts per microsecond (a thousand times too slow). Thus the electrostatic speaker must include an expensive and heavy step-up transformer which steps up the output voltage of the audio amplifier to a high voltage signal appropriate for adequate acoustic output from the electrostatic speaker. This expensive hardware raises the price of the speaker system. In addition, transformers have problems. At high frequencies they act like a large inductor in series with the amplifier, causing a large roll-off in the high frequency response. At low frequencies, the core saturates, causing high distortion and poor bass performance. Thus the designer is faced with the contradiction that for good high frequency performance he needs a small transformer with low inductance but for good low frequency performance he needs a large transformer so the core doesn't saturate. We know of only one manufacturer that has made a satisfactory solution to this problem and that is Acoustat, which uses two step-up transformers per speaker, a small and low inductance one for the highs, crossed over to a large transformer for the bass. 0f course this increases the cost and complexity of the system. Acoustat has been modifying their crossover parts ever since the system was first designed. To their credit, their field modifications are inexpensive and easy to do and all their old electrostatics can be updated to what they claim is correct now.

The transformer, even properly done, presents a mixed blessing. It does "step-up" the slew rate of the amplifier in proportion to its step-up ratio (usually times 100 or thereabouts) so the slew rate to the high voltage electrostatic panel is adequate. Unfortunately, it also "steps-up" the equivalent capacitive load the amplifier must drive. An electrostatic panel is actually a giant film capacitor with a "loose plate" in the middle. Typical panel capacitance is in the 500 to 1000 picofarad range. When you translate this load back thru the transformer (capacitance times 100), you are connecting your amplifier to a load that looks much like a .1 μF to 10 μF capacitor (depending on the electrostatic brand) directly across its output terminals. Consider that some amplifiers "blow up" when seeing this kind of load, and that most others generate substantial ringing and oscillations. A capacitor looks like a dead short to your amplifier, which its output stage must charge. Most output circuits current limit and slew into a typical electrostatic load. Obviously the output of the electrostatic speaker will be highly distorted if the amplifier driving it is going into overload. It is a rare amplifier design indeed that can drive an electrostatic speaker with good linearity (crass commercial comment – all our mos-fet amplifiers will, just fine) and you can throw IHF specifications out the window as amps are not specified under capacitive load test conditions (I doubt if a company would continue to advertise after it got a 100% distortion at zero watts rating in an audio magazine).

Further disadvantages. The full range panel has the same or nearly as much acoustic output from its back side as from its front. This output is reflected from nearby surfaces which may, or may not, be of sonic value (we suspect not) depending on the acoustics of your room. Typically the speaker should be located several feet away from any walls, and the rear wall should be damped to absorb unwanted back side output from the electrostatic panel. Some people claim the electrostatic, because of its large panel size, to be very directional, giving a small listening "sweet spot" in the room, making it a bit of a selfish speaker (if you are not in the "one person" sweet spot, it may not sound right. Many electrostatic speakers are fragile and can be permanently damaged due to electrical arcing between the elements if pushed too hard. (Acoustats are very rugged.) Some are weather sensitive and subject to more arcing in humid weather. (Don't use them on a damp concrete floor).

Bass performance of electrostatic panels is generally poor for four reasons: 1). The back waves come around and cancel the front waves at some low frequency depending on panel size (the bigger the panel the lower the frequency where bass cancellation occurs). 2). The elastic film diaphragm has a high "Q" at low frequencies and tends to "drum" and ring. 3). Most electrostats cannot handle much power and low frequency bass may drive the film into the grids, causing arcing and damage. 4). The step-up transformer may saturate at low frequencies causing muddy bass and distortion. Thus many electrostatic owners find themselves shopping for an electronic crossover, second power amplifier, and a pair of subwoofers soon after purchasing their speakers to attempt to get better bass response. This of course makes the real cost of electrostatic speakers much higher than originally planned. Our advice: Improvements in conventional speakers in recent years have eliminated many of their disadvantages in high frequency performance, and in most other respects, conventional dynamic speakers win hands down. We don't think electrostatics are your best choice for the money.

ELECTROMAGNETIC PLANAR SPEAKERS. (Magnepan and Infinity are type examples, although much different from each other.) In these speakers the diaphragm is driven directly by the power amplifier using a coil located in a permanent magnetic field. There are no step-up transformers or connections to AC power.

The Magnepan speaker is somewhat similar to a full range electrostatic (especially in appearance) being a flat panel the size of a door with a stretched film membrane giving acoustic output from both front and rear. The "voice coil" is a wire glued to the diaphragm running zig-zag up and down the membrane from top to bottom. The magnets are strips bonded to a perforated panel located in front of the diaphragm. AC current (signal) thru the voice coil attracts and repels the diaphragm in relation to the fixed magnets giving the acoustical output.

Advantages: Similar to an electrostatic in having a large sound field, elimination of "cone" breakup, elimination of "box" resonances, and back side output pleasing to some. Less costly than an electrostat as the power supply and step-up transformer are unnecessary. A much nicer amplifier load than an electrostatic. Very durable and no electrical "arcing" hazard.

Disadvantages: Similar bass restrictions as #1 and #2 mentioned regarding electrostatics, leading to the same bi-amp and sub-woofer costs, making the real cost higher than expected for those wanting extended deep bass-performance. Similar placement restrictions to avoid undesirable back wave reflections. Limited vertical dispersion, although true of electrostatic panels too. Most models are "single-ended" (magnet on only one side of voice coil – rather than being "push-pull" as are most electrostatics and conventional dynamic speakers). Since the field strength of a magnet falls off with the cube of the distance (when the distance to the magnet doubles, the field strength is one-eighth) the drive force to the diaphragm varies dynamically in use, a nonlinear mode of operation. This tends to (although not a gross problem) compress transients and dynamics and to some ears, restrict highs. Our advice: The lower priced models are good value for those who find the big sound field important (remember, no system is perfect) but we suggest that the lower priced Acoustats may be more desirable than the higher priced Magnepans if your amplifier can drive a capacitive load, and if you desire a planar type speaker system.

The Infinity EMIT and EMIN tweeter and mid-range panels could be considered to be similar to "miniature" versions of Magnepan type design, but with the voice coil being an etched film and with very powerful magnets. They are used for high and mid-range reproduction only in various combinations in a wide series of Infinity speakers, along with more conventional closed box design woofers. Outside of cost, we see no disadvantages to this engineering design and some consider the tiny EMIT and EMIN units to be of superior design.

HORN LOUDSPEAKER DESIGNS. Those systems using megaphone shaped horns in front of the driven acoustic diaphragm to increase the acoustic efficiency of the driver(s). (Type examples are Klipsch, some Altec, Frazier, JBL, and most "PA" speakers.)

In the mid-range and tweeter horn units, the diaphragm is usually small (an inch or less across). The acoustic megaphone will be tuned to the frequency range of desired acoustic amplification. Thus mid-range horns are much longer than tweeter horns. Horns for bass speakers (where the wavelength needing amplification is much longer) get very big and their use is not common (except for some Klipsch "folded" corner-horn designs in which the corners of the room become extensions of the bass horn. In England, some diehards build concrete bass horns extending far out into their back yards, or make use of old fireplaces and chimneys common in old British houses and redesign these as bass horns – with the woofer going where the chimney outlet used to be. (This makes it difficult to trade speaker systems.)

Design Advantages: Efficiency. The acoustic horn speaker system requires ten to one hundred times less amplifier power for equivalent sonic output. This means your 10 watt amplifier becomes a 100 to 1000 watt amplifier when used with horn loudspeakers. Since any amplifier becomes much more linear when driven only at low power, use of horn speakers kind of automatically gives you a "better" power amplifier. In P.A. applications covering auditoriums or open air theaters, horn speakers are required for the sound pressure levels desired by many rock groups. With more conventional speakers they would require ten times as many huge P.A. power amplifiers as the great stacks of them they already use, which would probably burn out the electric power lines feeding the place. Some claim that horn speakers exhibit less "doppler" distortion as the physical movement of the diaphragm is very little. They claim that a conventional cone, which may have as much as an inch of movement back and forth, will cause a pitch modulation of a high frequency wave that is being reproduced at the same time as a low frequency sound wave, causing a "flutter" tone in the high frequency sound wave. We suspect this is not a "worse case" problem.

Design disadvantages: Poor-signal to noise ratio. Since the acoustic horn amplifies everything from your electronics, the residual noise will be amplified too, and you may find that an amp and preamp that were quiet enough with conventional speakers will now have annoying background hiss or hum. The noise from the electronics has not changed, the horn speaker has amplified it. Limited frequency response. The acoustic output of a horn drops off very rapidly beyond the rather narrow range for which it was tuned. Even the biggest folded bass horns have little output below 50 Hz, and horn tweeters rarely get much beyond 12,000 Hz linearly. Ragged frequency response. Inasmuch as it is rare to find more than a 3-way horn system, the response dips and peaks (as much as 20 dB) going thru the audio range as the signal goes thru ranges where the horns are highly efficient and then areas where the efficiency has dropped way off. Non-musical acoustic resonances. The horns actually acquire their sonic efficiency by acting as undamped acoustic oscillators operating on the column of air in their throat. Undamped oscillators oscillate, and have acoustic output at their fundamental resonant frequency when there is input at harmonics of that frequency. You get output where there was no input. This contributes to "horn" or megaphone sound with many horn systems (in worst cases an annoying "honk" overtone). Spurious resonances from the horn structure itself contribute to this effect, and can be substantially reduced by damping as described in *Audio Basics*, Volume One, Number One. Size. Bass horns are big and won't fit in many rooms. The cost of bass horns is very high too.

Our recommendations: The disadvantages of horn speakers may, or may not, outweigh their efficiency advantages. In rational sized rooms, assuming a reasonably powerful amplifier, we suspect you will find that more conventional speakers are more linear and pleasing to you. If your requirement is loud, you probably should consider a horn system. Note too that the best horn systems (such as the big Klipschorn are fine speakers. They can, in a large listening room, driven with quality electronics, produce a "life-size" sound stage with amazingly life-like dynamic range. They are not perfect and their non-linearities are "different" from more conventional speakers, but no design is perfect, and that is why you have more than one kind of speaker to choose from (there is no "best" design).

PIEZO/CERAMIC SPEAKERS. Tweeters using a small solid crystal element instead of a magnet and voice coil. The crystal is bonded directly to the tweeter diaphragm. The electrical input from the amplifier causes the crystal to "twist" thus imparting motion, and acoustic output into the diaphragm. (Motorola piezo tweeter is a type sample).

Advantages: Much greater power handling than a conventional tweeter as there is no voice coil to burn out. Acoustic output rolls off at lower frequencies, simplifying crossover design in inexpensive speakers. Inexpensive and relatively easy to use and mount.

Disadvantages: Limited efficiency, thus many are supplied in crude plastic "horns" having similar spurious resonances as horn tweeters described above (actually worse, as the horn structure is very light plastic). Because the designer can "get by" without a low frequency crossover, many do, and although the output does roll off, it does not do so evenly, and thus the piezo tweeter tends not to be used in as linear a mode as could be done. Note that Motorola now has paralleled piezo tweeter units in a flush mount package that don't have or need the plastic "horn" throat. The support structure still has undesirable resonances.

Our recommendations: As supplied stock, piezo tweeters are too "ragged" for quality speaker systems, but, if the plastic support structure is carefully damped, they can work just fine, and may be better than conventional tweeters found in less expensive systems.

DIRECT-REFLECTING SYSTEMS. Those systems in which many of the speakers face the rear, in which the claim is made that the sound from the rear of the cabinet, reflecting off the walls behind the speaker, creates the illusion of "concert hall" acoustics. This claim is usually justified with the reasoning that "in a concert hall much of the sound you actually hear is reflected from the walls of the hall and is not direct sound."

A typical type example is the Bose 901 series loudspeaker.

Design advantages: This type of system can, if properly located, produce a large and pleasing sound field on some types of music.

Design disadvantages: One cannot "turn off" the large sound field on music inappropriate for this (such as a solo voice in a small room).

We would also suggest there may be flaws in the logic supporting the claims for the desirability of a system with substantial "reflected" output. One could point out that if we took the orchestra outside, and recorded it playing on an open flat field and then played the recording back in our home on a "reflecting" speaker system that created a "concert hall sound field," that we would then have a concert hall sound that did not exist. Although you may like this, it is a departure from the reality of the acoustic environment of the recording.

Another disadvantage is that many reflecting speakers, including the 901, are designed for active equalization. The raw system frequency response is down substantially at both high and low frequencies. The designers claim to overcome this by supplying with the system an active frequency equalization circuit, which, it is claimed, shapes the signal to the power amplifier to compensate for the raw response of the system, thus giving satisfactorily flat response.

The "catch" to this design technique is that for each 3 dB boost in acoustic output (an increase barely audible by most people) a doubling of amplifier power is required. Thus, if the boost at 20 and 20,000 Hz was in the area of 15 dB, for example, the amplifier would have to be thirty two times as powerful at these frequencies as in the mid-range where no boost was applied. Thus, if you were driving your amplifier to a 10 watt level on mid-range material, the equalizer would drive the amplifier to 320 watt levels at high and low frequencies, clearly beyond the capabilities of most amplifiers. Active equalization can only work well if the amount of boost is within the power capabilities of your amplifier. There ain't no such thing as a free lunch.

Finally, remember that microphones are stupid. The mike cannot distinguish between direct or reflected sound in the recording process. The microphone picks up all the sound at its location, no matter whether it came directly from the instrument or was first reflected from the recording surroundings. This sound, a mix of direct and reflected acoustics, is sent on to the recorder, and if the engineering is good, will finally show up on your record. In a linear system, the output of front facing speakers will play back this mix of direct and reflected sound (what the microphones "heard") in the proper proportion (assuming good judgement by the recording engineers in original microphone placement) and your system will play back the sound of the "hall" as it existed quite satisfactorily.

We would suggest that further deliberate "reflections" in your room is, at best, redundant. We prefer, in fact, a very "dead" and non-reflective listening environment in which we listen to the direct output of the speakers, with the acoustics of the listening room minimized. We desire to hear what is on the record, and only what is on the record. If the recording is bad, we want to know so, if it is great, we want to hear it, and nothing else. Fortunately, there are enough super recording engineers around (Jack Renner and Bob Woods of Telarc, for example) that our desires for well engineered records are met.

To be continued next month.

Frank Van Alstine

VOLUME TWO NUMBER THREE MARCH, 1983

We will continue with loudspeaker design and evaluation this month but first a couple of brief notes.

Troubleshooting revisited. Yes, many of you have written me that system troubleshooting is boring. Yet this month we received in a period of two days, three separate units (two preamps and one integrated amp) for "repair" because a channel was out. Two of the units came without advance notice and the other was sent by a previous troubleshooting contest winner who told us he had checked everything. In each of the three units the only problem was that the setting of the tape monitor switch had been disturbed (not firmly engaged in the "input" position) and thus a channel was open at this switch contact. In each case the "repair" was a one second push of the finger to set the tape monitor switch into the input position. Aaaarg! Please read our troubleshooting guide again. Please check everything before returning equipment for repair. Please call first, 90% of all "problems" can be resolved over the telephone without returning the equipment.

Although I have not used much of this space for crass commercial purposes (you have paid for information, not advertising), I would like to tell you about a new product we have developed that might be of value to many of you. It is our new mos-fet amplifier circuits for the Dyna SCA-50 integrated amplifier. The SCA-50 was a really fine idea and layout. A compact integrated full function amp – preamp at a low cost. It has switchable tone controls, tape to tape monitoring, all the "creature comforts" one could ask for. Many thought it was the best Dyna preamp section ever done. Its problem was it wasn't reliable – it often blew outputs for no good reason. Actually there were three reasons: 1. The compensation of the preamp ICs was badly done, leaving the phono and line sections marginally unstable and ready to turn into oscillators at any time. 2. The flow solder work on the PC cards was dismal, parts actually fell off in shipping. 3. The worst "bung" was the power amp design. It used plastic output transistors with a 60 volt rating. The power supply rail to rail voltage is 70 volts. Thus, at clipping, the outputs are overvoltaged and in the ready to blow-up mode, which they often did. The heatsink was designed for the plastic TIP outputs only and there are no higher voltage devices available with an adequate current rating, thus, for a long time we did not work on SCA-50s as there was no way to make them reliable.

However, it was just too nice a layout to permanently pass up and we have recently designed a whole new power mos-fet amplifier section for it. It includes new cast heatsinks, new audio cards (a miniature version of our MOS-FET 150B circuits complete with matched pair front end devices), and the same rugged Hitachi 2SK135 - 2SJ50 power mosfets we use in all our bigger amplifiers. They cannot be damaged in this circuit application. We increased the preamp power supply from 200 µF to 10,000 µF, installed new preamp ICs and re-engineered the compensation so the preamp is now absolutely stable. It is now one real honey of an amplifier, one we suspect many will feel is nicer sounding than many $2000 amp - preamp combinations. It puts out 30 real watts per channel (you will think its 100 watts) and the cost for a complete new unit is $350.00. We can rebuild any existing SCA-50 for $200.00. Stereo Cost Cutters is selling unassembled SCA-50 kits for just $88.00 (we will charge you $50.00 extra for assembly of these) so this may be a real opportunity for some of you that have been looking for audiophile grade equipment but not audiophile prices to obtain a real "keeper" of a control amplifier cheap. It will accept a digital disc player without overload too. We are going into continuous production of this unit and will have many available. Tell your friends about it.

BACK TO LOUDSPEAKER SYSTEM EVALUATION:

BASS RESPONSE seems to be one of the most misunderstood characteristics of an audio system. Everybody wants clean, deep bass, but few know what it really is, and fewer know how to go about getting it. The search for bass is expensive, there are all kinds of add-on components out there claiming to provide you with "better" bass. Many go off into the never-never land of adding electronic crossovers, bass amplifiers, and subwoofers without ever asking themselves the most important question first: Does my phono cartridge, turntable, preamp, and power amplifier really play deep clean bass now? If they do not, and most do not, then all the expensive equipment added downstream will never provide you with "better" bass, it will only play low frequency garbage louder (which, by the way, is what many audiophiles think is deep bass).

We better first attempt to explain what deep bass response of an audio system is (or can be if done properly). It is the sound of the low frequency characteristics of the acoustical musical instrument. Please try and remember that drums do not go "boom," organs do not go "blat," that each note of a string bass is a different pitch and tone. Note, that you cannot tell what the sound of an electric guitar is, as what you hear has been compromised by the electric pickup, cruddy P.A. amplifier, and scuzzy P.A. speaker system. Remember that each organ pipe has its own location in space, as does each individual drum in a drum set. The sound of musical bass is complex. The organ pipe turns on with a "chuff" sound, followed by the first tone and shortly thereafter by the sustained complex tone as the entire pipe starts working. It does not sound like a sine wave or square wave. Each pipe in each organ has a slightly different harmonic characteristic. Each drum is a bit different (many "modern" plastic skinned ones are pretty bad – they sound like a "hi-fi" set). With a big drum you can distinctly hear first the stick hitting the drum, the first tone of the skin, followed shortly by the sustaining sound of the drum as a whole. If, on your system, drums go "boom – boom – boom," you are not hearing drums, you are hearing the sound of your hi-fi system.

An interesting characteristic of obtaining accurate deep bass in an audio system is that most of the time you will perceive less bass. Most records (almost all modern "pop" and "rock" records) have nothing below 80 – 100 Hz recorded on them. An interesting aside is that probably none have any information above 12,000 Hz either – but that is another story. Note that "half-speed" mastering makes it much more difficult to capture deep bass cleanly, as at very low frequencies the cutting equipment must be linear to a much lower frequency than normal. A 30 Hz tone, for example, would have to be cut at 15 Hz, beyond the linear range of normal tape machines, cutter amps, and cutter heads.

When you do obtain truly clean deep bass, on those records that do extend into the "foundation" levels (Telarc digitals, many well engineered direct discs, some re-masters, and an occasional competently engineered standard recording) the result is a solid "foundation" (more sensed than actually heard) to all of the low frequency material. The drum, the organ, the string bass, the big brass instruments, the entire orchestra, all take on a greater sense of reality. "Telarc" bass (with the exception of those impossible cannons) is never "too loud" – those audiophile magazines that have criticized Telarc bass as being overdone are simply describing the underdamped distortion their system adds when attempting to play the information. Their system is bung and they don't know it.

What is necessary from an engineering standpoint for clean bass? Let's set forth a few basic rules and observations. (We need to know what is required before discussing how various loudspeaker designs achieve – or don't achieve – these goals.)

For accurate bass response, every component in the audio system must have a damped Q. Oops, another question to answer, what is "Q"?

Crudely, "Q" can be defined as the ratio between stored energy and dissipated energy in a mechanical or electrical system. (Note that the term "damping factor" could be considered to be the reciprocal of "Q".) A few examples:

If one had a "perfect" spring in which there was no internal dissipation of energy, then an input into that spring would set it "bouncing" forever. The system Q would be infinite.

In contrast, a totally non-springy material (PlaSt-i-clay comes close) will, when an energy input is made, simply deform, absorbing all the input and dissipating the energy, and remaining in the deformed state with no return at all. This would represent a Q of zero (or an infinite "damping factor").

It isn't quite that simple, there are "fudge factors" used mathematically to compute the value of Q, but conceptually the fudge factors do not need to be considered.

Obviously, we do not want a Q of zero in our audio electronics or mechanical circuits or there would be no output at all. Neither do we want an infinite Q, or the first note in, at the resonant frequency, would continue to play forever - boooong.

What we want is a circuit in which there is adequate linear output to cover the audio bandwidth of interest, but without excess underdamped oscillations at the frequency extremes. This condition is called "maximally flat" damping, or a "Q" of .707. Another condition, called "critically damped," is achieved with a "Q" of .5. This is the maximum bandwidth condition you can have with no resonances at all, although discussing the relative merits of a Q of anywhere between .5 and .8 is splitting hairs. What is to be avoided at all costs is a higher Q, in which the system turns into an electrical and or mechanical spring at low and high frequencies, having an oscillating output when there is no input.

In a loudspeaker system, a cheap, sneaky, and conceptually improper way to get "better bass response" is to tune the woofer, crossover, and cabinet to have a high Q at some low frequency. This will provide the manufacturer with a much better bass "specification" as the under-damped oscillator (that should have been a woofer) will have substantial acoustic output at the frequency at which it has been tuned (just as your car will have substantial up and down motion when the shocks go out). Unfortunately, the woofer can no longer start and stop quickly (it keeps oscillating when the signal is removed) and has output at the tuned low frequency with an input of a transient signal too. If your speaker is "tuned" for "more bass" at 30 Hz, for example, it will put out underdamped oscillations at 30 Hz. It will also put out underdamped 30 Hz oscillations with a transient input. It isn't a bass reproducer, it is a low frequency noise maker. This isn't music, this is distortion! A good clue to look for in a speaker test or specification sheet is the low frequency distortion. Typically it will be over 20% and increasing at higher power levels. A well damped speaker will have less than 5% distortion at low frequencies, and really fine ones less than 2%. An honest speaker manufacturer will tell you the system Q, and it must be less than .8 or the system cannot play bass, it can only make its own "bass." More on loudspeaker cabinet design later.

The point of discussing "Q" with you is to point out that an underdamped "Q" can occur anywhere in your system, not just in the loudspeakers, and with the same non-musical effect. A few examples:

Most phono cartridges have underdamped bass resonances. In many, as discussed last year in *Audio Basics*, the entire stylus assembly is not solidly located in relation to the body, causing underdamped low frequency resonances. The stylus suspension also has underdamped resonances of its own, and the entire assembly, mounted in the tonearm, generates further underdamped resonances. Your system cannot play bass if the cartridge – arm cannot play bass in the first place. If your "bass" isn't "good enough" look to your phono cartridge first! Note that our $99 Longhorn Grado is damped and will help many systems play bass, not "make bass." It may be of more value to you than a sub-woofer.

Many RIAA phono preamplifiers cannot play bass. For example, in vacuum tube preamplifiers, even the most expensive and exotic, there just is not enough open loop gain to follow the RIAA equalization curve at low frequencies. Below 500 Hz the response falls apart under dynamic conditions. If you examine the RIAA equalization parts values in many vacuum tube preamps (ARC units, for example), you will find the values have been "tweaked" to get much more bass output than the standard calls for. This to try and make up for the gross low frequency non-linearities that occur when the stage runs out of feedback control. Unfortunately, "bumping" the equalization curve to give "more" bass only introduces underdamped resonances and phase errors into the system. If this is the "warm concert hall bass you love" well, we are sorry, for what you are "loving" is distortion, and noise never made by any acoustical musical instrument. Of course you have the right to "like" this distortion, just don't try and argue that it is "good." Bad taste knows no bounds.

Many designers (with the approval of underground audiophile flakes) have developed the hairbrained idea that "to have good bass" you have got to be "flat to DC" and have gone into the engineering never-never land of attempting to design totally DC coupled amps and preamps from input to output. Lots of luck, it cannot be done. Why not?

1. There is no such thing as an ideal power supply – one that can supply constant voltage under any load condition. At subsonic frequencies, the ripple on the supply goes up and the voltage drops. Thus the distortion skyrockets and usually the interaction of the supply and audio circuits causes an underdamped high Q situation. Does your amp or preamp go "wop – boom" at turn-on or turn-off (or require an output relay to prevent this)? Congratulations, you have a high Q at low frequencies, your audio electronics are underdamped resonators and you have low frequency noise generators, not bass reproducers. A sub-woofer is not going to help you, unless you like the noise louder yet.

2. No real world active device maintains linearity as you approach DC signal levels. Internal thermal related distortion rises and the linearity of the circuit falls apart. This is the reason the distortion curves rapidly rise at low frequencies, and why in IHF tests, measurements are cut off at 20 Hz. You wouldn't want to see the distortion in your equipment at 2 Hz, for example, where your .001% THD specification has skyrocketed to 50% THD!

3. There are many sources of signal at near DC levels you are not going to like very much. AC power line voltage changes, record warps, and turntable rumble, for example. It is not a clever idea to let your amp or preamp accept these signals and attempt to amplify them to full power. We engineer audio amplifiers and let the flakes build the AC power line level detectors. Your speakers will like our audio amplifiers better.

An interesting "destruct test" for DC stability in your electronics is the Telarc 1812 cannons. Although this record will clip any amplifier, how the amp clips is revealing. In many cases, the amp and/or woofers will blow up when this cut is played loud on a big "DC" amplifier, as the clipping drives the unstable amp to the rails offset. A DC stable amplifier will clip, but without centerline offset or drift, and not damage itself or the speakers.

To have stable and linear "bass" performance, the electronics must be bandwidth limited so as to never accept a signal outside of the range in which the internal devices, and power supply, can operate linearly. You cannot stuff ten pounds in a one pound sack. Good perceived musical bass performance is not obtained by DC signal level acceptance, it is obtained by accepting a signal level only within the internal capabilities of the circuit. If you have a "DC" amp, you can never obtain clean musical bass performance. Don't worry about which sub-woofers to buy, that isn't where your problems are.

Another implication of underdamped Q making noise, not music, is the use of an equalizer or equalized speaker or "electronic sub-woofer" in your system. The way a large bass boost is generated electrically is with an underdamped high Q resonant circuit! Whoops, boosting bass electrically brings you right back to square one – you now have under-damped low frequency noise, not better bass. You cannot win. The only rational use for an equalizer is to tame room resonances. A square room may, for example, have its own tuned resonances around 80 Hz which will make the system boom no matter what you do. The only cure may be to move, or to use an equalizer to cut the level at the resonant frequency so you can stand to live with the sound. (In general, moving is a better choice.)

Again, before even starting to think about buying "better bass" in the speaker systems, be very sure the equipment ahead of your speakers can actually reproduce bass. The chances are very high that they cannot.

I suggest one final test. Acquire a copy of the M&K RealTime record, *Flamenco Fever*. The third cut on side two, *Hands and Feet*, is a "killer" musical test of bass transient response in your system. It consists of a flamenco dancer accompanied by a large, powerful drum set. In the middle of the cut the stomping and drumming get going at a frantic pace. In almost every system I have heard this play on, the bass goes "berserk", turning into a rubbery mess, indicating awful underdamped resonances. When the system is competent, each individual drum beat starts and stops, none running together. Each footfall produces the sound of the stomp, together with the powerful, but nearly sub-audible, sound of the floor being stomped. The drums have enormous sub-sonic power and tone structure, but no "boom" at all. The dancer and the drums never blur together. It is an outstandingly well recorded and engineered record, and far beyond the playback capabilities of most systems. It does not take a group of audiophiles setting around for hours agonizing over subtle differences in balance and harmonics trying to decide which is "better." The results with this record regarding bass performance take no listening skill or experience at all, it will either be musically "right" or it will be an awful muddy mess. We dare you to try this record on your system and report back the results to us. You may be unpleasantly surprised. Remember, if the bass runs together and becomes muddy on this record, it is doing the same on all other records too, its just that this record lets you focus on this aspect of the system performance much better than most.

O.K., you are sure your cartridge and electronics really work well, so lets go ahead with the engineering of good bass in a speaker system.

WHAT ABOUT SUB-WOOFERS?

The idea and goals of sub-woofer use are just fine. The object is to extend the deep bass response of your system by adding an additional speaker, or speakers, that are specifically designed to reproduce deep bass only. The sub-woofer is used with an additional crossover (either passive or active) whose job is to see that only the deep bass signal is fed to the sub-woofer, and usually to cut off the deep bass to your main speaker. This is done because your main speaker may be non-linear at very low frequencies or be fragile and easy to blow up on heavy powerful bass material. If the crossover is passive (the less expensive way to add sub-woofers) it may be built into the woofer cabinets themselves or be supplied as a separate black box. The speaker outputs from your power amplifier are connected into the passive crossover, which splits the signal and sends the deep bass to the sub-woofer and the rest to your main speakers. An electronic crossover is a more expensive proposition. In this case, the signal is split between the deep bass and the rest at line level, after the preamplifier. The preamp feeds the full range signal into the electronic crossover "black box." The electronic crossover will usually have two sets of outputs, low or deep bass, and high, the rest of the range. The segregated signals are then fed into two stereo power amplifiers, one to drive the sub-woofers and another to drive the main speakers. This system is called bi-amplification and although usually done to add sub-woofers, can also be used to split the signal in the mid-range area too. More on the electronics of bi-amplification later.

Obviously the use of an electronic crossover and bi-amplification is a very expensive way to get better bass. You must add the total cost of the electronic crossover, the sub-woofers themselves, and a second high quality stereo power amplifier, and compare this cost with the cost of trading in your speakers for simply a much better full range speaker system that in itself has the deep bass response you desire. We would suggest, all other things being equal, that the cost of a better full range speaker will be less than the cost of bi-amping and adding sub-woofers.

Although using sub-woofers with a passive (after the power amplifier) crossover is less expensive, it does negate some of the good engineering reasons for use of the sub-woofer, which are as follows:

1 It is generally accepted that the less "work" an amplifier has to do, the more linear it will be. Thus, if you can split the signal load between two amplifiers, each will work better. The bass amplifier, not having to contend with high frequency slew rates, should play bass more linearly. The high frequency amp, not having to handle the large power supply draw from heavy bass material, should play highs more linearly.

2. In theory, the use of two identical amplifiers in a bi-amped system will give you four times as much power. For example, if you had a 100 watt amplifier (28 volts RMS output) then a 28 volt RMS high frequency sine wave would be its maximum output before clipping. If you wished to play a 28 volt RMS low frequency signal at the same time, obviously the amp would be driven far into hard clipping (28 volts + 28 volts = 56 volts desired output and only 28 volt capability). However if you have two 100 watt amps and play the 28 volt high frequency signal with one, and the 28 volt low frequency signal with the other, then you do get a total of 56 volts RMS from both without either clipping and that is 400 watts of total power, not 200 watts.

3. Many think that loudspeaker crossovers, containing large coils, capacitors, and resistors are a difficult load for an amplifier to drive. In addition, various drivers are padded down with resistive loads to balance the system. Much of the amplifier power may be wasted in heating resistors, not in acoustic output. In a bi-amp setup, it may be possible to eliminate the passive speaker crossover, set the balance ahead of the power amplifiers, and get full useable power directly into the speakers.

However, TANSTAAFL! (There ain't no such thing as a free lunch). In practice, there are many pitfalls involved with bi-amping that in our opinion, more than negate the possible advantages discussed above.

1. You are adding the distortion of the electronic crossover to the system. It is likely that the distortion added is more than the distortion reduced in the main amplifier by easing its load, especially if the main amp is a very linear design.

2. You are making the presumption you can "graft" together two different speaker systems from probably two different sources and get a smooth crossover shift between them – that you are a better loudspeaker engineer than that employed by the full range speaker manufacturer. This is possible, but not likely.

3 Do not even assume you can adjust the levels between bass and top with any degree of accuracy. The Dahlquist LP-1 is about as good as electronic crossovers come, but its crossover and levels are set with front panel knobs that are not exactly calibrated. You don't really think that just because you have the pointer on the knob set at

"80" on the faceplate, for example, that you really have an 80 Hz crossover point, do you? You must have a dual trace oscilloscope and a signal generator to set the dials to obtain exactly the same crossover point and level on each channel, a requirement if you desire both sides of your system to be the same and to have adequate imaging characteristics.

4. Unless the input impedances of the amplifiers you use remain constant under dynamic conditions (most do not) the two amps used will interact with each other affecting both the crossover points and the system response and distortion. It is possible to try a new bass amp in a bi-amped system and find that you have changed the sound of the highs, a very neurosing situation.

5. Finally, do not assume that a sub-woofer really will reproduce deep bass. In general, sub-woofers are big woofers in little boxes. The chances are it will be a high Q system, tuned for lots of "output" in the 20 - 40 Hz range. It is likely to just be another underdamped resonator changing your bass noise spectrum to a lower set of frequencies. Before buying a sub-woofer see if data is given for its distortion at full power and for the system Q. We suspect you will be shown data only claiming some deep frequency response, without mentioning system Q or distortion at all. Low frequency noise isn't music.

Actually deep bass is very expensive to obtain. It requires a big and high quality woofer in a big box. The construction must be rugged to keep the box from having more output than the woofer. Both KEF and B&W have some very interesting laser interferometer data showing speaker cabinets acting like big "balloons" when driven hard at very low frequencies. Construction, handling, materials, shipping, and sales overhead costs rapidly rise in dealing with big, heavy, honest woofer systems. Expect to pay $2000 a pair for systems that will make an honest 50 Hz with low distortion and $3000 a pair to make 40 Hz. If you want an honest flat 20 Hz, sorry, you can't have it, unless you can get Dr. Richard Griner of the University of Wisconsin electrical engineering department to tell you how he made his systems. Of course he was using ten 200 watt per channel amps to drive the whole mess the last I knew.

To be continued.

Frank Van Alstine

VOLUME TWO NUMBER FOUR APRIL, 1983

Before continuing with loudspeaker evaluation, we have a couple of short subjects to cover this month. A client recently sent me a letter describing a strange problem with his system that I had not considered, and that may save some of you much service time and trouble. We had built this client a MOS-FET 120 series amplifier, and he claimed the power transformer "hummed." The unit was returned twice to us for warranty service, but it never hummed here, and we could not find any problem, and had run out of ideas of what could possibly be the cause of the problem. The client had even gone so far as to locate another power transformer for the unit and had it installed by a local serviceman. The unit still hummed. At this point the client and I were both starting to think that each other was crazy. The following letter from the client finally describes the solution to the problem:

> "Dear Frank: Thank you for your suggestions regarding the pursuit of my problem with my MOS-FET 120. The problem turned out not to be with the 120 or its transformer. Rather, I investigated whether it was simply a problem with the voltage (AC power line) coming to the amp and, indeed, discovered that some equipment on the same circuit was causing the problem (the awful transformer hum). What was particularly maddening, as well as a relief, was that the culprit was in my own bedroom. It turns out that my waterbed heater, when running, causes the buzz. If I hadn't spent $100 and many hours of searching for a new transformer, etc., in attempts to fix the 120, it would have been a funny end to the episode. Once I switched the plug to an outlet on the other side of the living room, the hum lessened considerably, and I'm going to insert a filter between the outlet and the amp, to see if I can decrease it still further. Even if it doesn't, I now know where the problem is, so I can take other steps to eliminate what is now a very tolerable hum. (Ed. note – first step should be to see if the waterbed heater is defective – it shouldn't put garbage on the AC power line if working properly.)
>
> Besides informing you of the outcome of the problem, I wanted to say two things in the letter. First, be ready to advise others that they should check their power source, should a similar problem be reported to you. Although my problem may have been highly unusual (or obvious?), neither my local technician nor I thought that the nature of the AC line current might be the problem. Had any of us thought to investigate this, much effort and money would have been saved. Second, I'm wondering if you know where I could sell an extra St-120 transformer?"

The second "short subject" is a do-it-yourself construction project for the Dynaco PAS series vacuum tube preamplifier (it also applies to our SUPER-PAS preamps) and may be of value with any vacuum tube preamp with a similar "problem." For those of you not directly affected, read about the project anyway, its kind of interesting to understand what simple things can affect the sound of your audio system – one of many conditions adding up to sound reproduction that falls short of reality. We still have lots of engineering to do before reaching our goal of good enough to be indistinguishable from real.

Would you believe that the pilot light (#53 light bulb) degrades the performance of the PAS series preamp? There is nothing "wrong" with the light bulb, it is not "a bad sounding part," nor is there a "just wonderful sounding magic light bulb" available that "sounds better." The problem is that in this application, the #53 light bulb degrades the overall quality of the PAS, and that a different kind of part eliminates this problem in this application, a part not invented at the time the PAS was originally designed.

The problem is one we (and others) have overlooked for years. The #53 light bulb draws too much current from the power supply! It draws about 140 milliamps of current in operation, and much more at turn-on (1000 milliamps = one ampere). Each 12AX7 or 5751 tube draws only 70 – 80 milliamps of current from the heater supply. Thus the light bulb is stealing as much energy as two of the tubes. Because the power transformer in the PAS is small, its energy capability is limited, and the current draw from the light bulb reduces its ability to supply the demands of the circuit. If the light bulb is removed, the ripple on the heater supply decreases about 20% (the same effect as putting in bigger heater supply capacitors), hum in the circuit is reduced, the heater supply DC voltages go up slightly, turning on the signal tubes a bit harder, improving their gain, reducing the loop distortion, and finally making a sonic improvement you can hear. All that because of a light bulb! The obvious and easy do-it-yourself project is to simply remove the light bulb. However, now you don't have a pilot light at all, which is annoying. Thus, the following project is how to change the pilot lamp easily to a LED, which only draws 25 milliamp of current, does not pull excess current at turn on (which improves transformer reliability), which is DC powered (eliminating an AC wire run which added noise), and which brings your PAS into the modern "solid state" age with a glowing LED on the front panel.

You will need the following parts from Radio Shack: One 1000 ohm, two watt resistor, one #276-041 T – 1 3/4 Jumbo diffused lens red LED (pronounced led RED in Japanese), and one terminal strip with two ungrounded lugs and a center lug for mounting it on the long #8 screw protruding up in the chassis from the heater supply parts stack. The spacing of the two ungrounded lugs should be adequate to mount the two watt resistor between them, one lead to each lug.

Remove the preamp from the system, remove the top and bottom cover, and turn the unit upside down on your bench. Remove the pilot light bulb from its socket. Remove the amber lens from the faceplate. The LED should be a press-in fit in the faceplate hole. If not, enlarge the hole or go back to Radio Shack for a slightly smaller LED. If its a loose fit, a bit of 5 minute epoxy on the rear side will affix it firmly. Remove the twisted pair of wires from the original lamp socket to the 12X4 power supply tube socket (pins 3 & 4 on the 12X4 tube socket). Leave the other pair of wires attached to pins 3 & 4 as they provide heater power to the tube.

Now turn the unit right side up and locate the long #8 screw protruding up from the heater power supply stack. In a stock PAS this screw will be holding down the two 2000 µF supply caps, along with the flat and square (green or black) selenium rectifier. In a SUPER-PAS, the stack will be two bigger capacitors, a single solder lug, and the supply diodes. Install the new terminal strip on top of this mess with another #8 nut and lockwasher, so the ungrounded terminals can't touch anything.

Install the 1000 ohm two watt resistor from one ungrounded lug to the other. Install a wire from one end of the resistor to the + side of the heater circuit. (In a stock PAS this is the red lug on the selenium rectifier.) (In the SUPER-PAS the large supply capacitor, 4700 µF at 16 volt or 6800 µF at 16 volt, + lead that is common with the banded end of one of the two blue or black diodes.) Now twist together a red and a black wire pair, long enough to reach from the new terminal strip thru the chassis hole next to the heater supply, and along the back and side under the chassis to the LED leads. At the heater supply connect the red wire to the other end of the 1000 ohm 2 watt resistor (the end not connected to the + heat heater supply). Connect the black wire to the – heater supply (in the stock PAS, the black lug on the selenium rectifier) (in the SUPER-PAS the – lead of the other large capacitor which is common with the unbanded end of the other diode).

At the LED, connect the red wire to + (anode) lead of the LED and connect the black wire to the - (cathode) lead. The cathode lead is adjacent to the flat side of the LED base. If in doubt ask the store where you bought the LED to identify the leads for you. Make sure the soldered connections cannot touch the chassis, the bottom cover, or anything else. That is all there is to it. The circuit is simple: From the + 12 volt DC heater supply thru a 1000 ohm resistor in series with the LED (to limit current through it) and back to the -12 volt supply.

Turn the unit on, the LED should glow red. If it does not, turn the unit off immediately and check your wiring. You probably wired in the LED backwards (which destroyed it). If in doubt, don't do the project or call us for help. All SUPER-PAS preamps as of this date will have the LED pilot lamp as standard at no additional cost. We will retrofit your SUPER-PAS for $20.00 if you wish to return it to us, but you should be able to do it yourself as described above.

Now, on with Loudspeaker Evaluation, part three.

There are many different kinds of loudspeaker cabinet designs for attempting to obtain clean deep bass response. Many claims are made, and the virtues of various cabinet designs have been long argued among audiophiles. The object of any of these designs is to prevent the sound waves from the back side of the loudspeaker cone from canceling that from the front side, and if done carefully, to "tune" the loudspeaker for the flattest response and lowest distortion. Any of the following designs can work well, if properly executed. All can yield terrible results if improperly engineered or if deliberately tuned for a high Q large bass peak. Some designs are more expensive to execute, some may trade off speaker efficiency for cabinet size, and all other things being equal, deeper clean bass response requires a bigger, heavier, stiffer, and more expensive cabinet. In the final analysis, it is the execution of the design rather than the absolute type of design that decides the linearity of the system. Your goal is to find a linear speaker system, not one of some specific design type. Don't lose sight of the forest for the trees. Some standard cabinet designs are as follows:

INFINITE BAFFLE: An ideal infinite baffle design would be a very large wall with the speaker mounted in it. Obviously the back wave could never "come around" to cancel the front wave. The volume of air both behind and in front of the speaker should be very large, thus loading both sides of the speaker cone equally. Assuming a very large infinite baffle (the Great Wall of China, for example, but extended straight up a few miles too) then the bass response of the system would be exactly that of the front side of the loudspeaker cone, for better or for worse. If the loudspeaker was in itself carefully damped (correct magnet size, correct voice coil, correct surround and suspension material for this application) then it would be possible to have a critically damped system and clean, undistorted bass response. If the loudspeaker itself has resonant peaks then they will still be there and the system response will still be poor. An approximation of an infinite baffle can be made by mounting the loudspeaker in a very large box in which the compression of the air in the box has negligible effect on the speaker cone. A good example of a reasonable infinite baffle cabinet is the older Bozak Concert Grand loudspeakers, a credible design 20 years ago and now. The disadvantages of the infinite baffle cabinet are that they are large and expensive, and there is no opportunity for the designer to use the air mass in the box to tune the loudspeaker for better response characteristics – what the loudspeaker does is what you get.

ACOUSTIC SUSPENSION: A smaller sealed box design, in which the compression of the air inside the box is used as part of the loudspeaker "suspension" In general the compliance of the suspension material of the loudspeaker will become greater (less "springy") as the air in the box becomes part of the "spring." Inasmuch as there still ain't no such thing as a free lunch, as the cabinet size is reduced, either the frequency response, the efficiency, or the Q of the system will be degraded. If the designer chooses to maintain a damped Q and desires the same deep frequency response as in an equivalent infinite baffle, the trade off to bring the box size down to what you can afford and to what will fit in your sound room is reduced system efficiency – you will need a more powerful amplifier. If this trade off is carried too far, you can end up with the contradiction that adequate power for normal acoustic output will also be enough power to easily fry the woofer voice coils, a major problem with some "mini-monitor" systems. (B&W prevents this last problem from occurring in their little DM12 by adding internal electronic protection circuits that prevent the system from being overpowered). It is as easy to tune an acoustic suspension system for a high Q and a big bass "boom" as it is to make it flat, but less efficient. Thus the specification for "Q" is something to look for when considering purchase. It should be .8 or less or you will be buying a boom box.

PORTED SYSTEMS: A loudspeaker system with a specifically designed hole(s) in the cabinet which if properly engineered can reduce the system Q, increase the efficiency, and/or extend the bass response. The "hole" can be either a simple hole, a tunnel into the box (tuned port), or a "passive radiator" (which is simply a loudspeaker without a magnet or drive mechanism who's diaphragm mass would be equivalent to the mass of air in a very long port tunnel). "Bass reflex" is a term sometimes used to describe a ported system. The advantages of this kind of system can be greater efficiency and/or deeper clean bass response than in an equivalent size acoustic suspension speaker. The disadvantage is that it is easy to design an even worse system with many large peaks. Many very bad inexpensive speakers are poorly executed ported designs, which have tended to give this technique a bad reputation. Another disadvantage is that the port can be a path to the outside for the inner box reflections which should have been damped and suppressed by the cabinet. Thus many ported systems exhibit a midrange "honk" from the non-linear higher frequencies "squirting" out the port. These can of course be suppressed by proper acoustic damping of the port. The passive radiator approach suppresses these reflections better than most. Again, the most important "specification" is system Q. It will be low in a properly engineered system and high in a boom box.

TRANSMISSION LINE: This design could be considered a special kind of ported system in which there is a long (sometimes folded) tunnel behind the speaker. If carefully designed, it is possible to couple the air mass in the tunnel to the speaker cone in such a way as to reduce the Q of the system so that either efficiency and/or frequency response is extended. This is an expensive design approach as the internal cabinet construction is very complex, and the box becomes very heavy. Sometimes it is possible to get "too much" bass using this design, with the woofers destroying themselves playing record warps, footsteps, and DC level shifts from unstable amplifiers. Some transmission line manufacturers recommend use of the "rumble filter" in the preamp to avoid these problems. This presents the strange contradiction to the user of having paid a bunch more money for deep bass, and then cutting it off in the preamp. Obviously, the speaker with a higher cutoff frequency and a damped Q becomes its own "rumble filter" but is less expensive. Why then pay for bass you filter out? Remember it is possible to tune a transmission line for a underdamped high Q "boom box" condition too. Again, examine the important specifications before you buy.

There are a number of variations on the above designs, with woofers facing down, ports on the back, etc. In general they are nothing new or magic, and will behave in accordance with the basic rules that you can manipulate the system Q, the system efficiency, and the system frequency response and that in larger cabinets you have higher efficiency and/or wider response while maintaining a damped Q. In dealing with a tuned resonant system (a loudspeaker moving air to produce sound), there is no magic method and we are not awaiting new scientific break throughs. It is simply a matter of execution of a competent design. By the way, don't hold your breath awaiting a "massless" driver with perfect response. If the driver has zero mass, then it can exert no force at all on the surrounding air. As soon as it has any mass, the same old rules of physics apply and you are back to dealing with just another tuned resonant system.

So much for basic loudspeaker design (at last!). Actually, much of what you hear in a loudspeaker (in addition to the music) is more determined by the execution of the system, rather than the kind of design. Ragged frequency response (large peaks and dips) are caused by sloppy design execution. Usually this is caused by using drivers outside the range that they are reasonably linear or by using a crossover design that is inadequate for the drivers selected. The most advanced speaker engineering companies (B&W, KEF, Celestion) know almost exactly what the response of their raw speakers are and have designed crossovers to nearly perfectly match the drivers, achieving seamless response. Many others design crossovers using faith instead of engineering, hoping the drivers are nearly flat and that standard crossover slopes will be adequate (they usually are not).

The other major sources of loudspeaker distortion are as follows:

SPURIOUS RESONANCES. The vibrations from the speaker diaphragm are transmitted into the speaker framework and into the support structure and cabinet itself. (See *Audio Basics* Volume one, Number One, January, 1982.) Only the speaker diaphragm should have acoustical output. The rest of the system should be inert. The most noticeable sonic effect of spurious resonances are what we call "vowel tone colorations." These give the sound an overall tonal cast somewhat like pronouncing one or more of the letters E, A, U, O, and I. Say aaaaah, or eeeeeeh. I suspect you can remember hearing this sound in some speakers while they were trying to play music. It is wrong.

DRIVER DIAPHRAGM NON-LINEARITIES. The speaker cone itself does not behave like the "perfect piston" it is supposed to under dynamic, or even sine wave conditions. We suspect Celestion has documented this better than most and you should obtain a Celestion SL-6 brochure which has many pictures of driver non-linearities made with a laser interferometer. All other things being equal, the speaker manufacturer that gives evidence of being able to document these problems probably has a better handle on resolving them than the companies that seem to be ignorant of their existence. Read their literature carefully. Do they seem to know about the major problems, or do they just claim their speakers are "wonderful" with no supporting evidence at all?

CABINET SURFACE REFLECTIONS. The sound from the drivers is adversely affected by being reflected from the front surfaces of the cabinet and from any protruding cabinet edges. It is also affected by heavy grill cloth framework supports. This problem is minimized by making the mounting panel as small in area as practical and many of the quality systems are quite narrow. A tweeter mounted on a big wide speaker face isn't a good idea. Carrying this to a logical extreme, B&W mounts the tweeter in free air on the DM7/II, 802F and 801F and even the mid-range is in a specially shaped cabinet with nearly no reflective front surface in the 802F and 801F models. In addition, these mid-range cabinets are made of concrete. Dahlquist mounts the mid-range and high frequency drivers nearly in free air on small panels in the DQ-10.

MAGNETIC MOTOR DRIVE NON-LINEARITIES. The speaker is a linear (as opposed to rotary) electric motor. Unless the magnet, coil, and their support structure are carefully designed for the required application major drive non-linearities can occur, such as: driving the coil and diaphragm out of the magnet causing loss of control and in the worst case, bottoming and damage. Slewing can occur in the speaker as, for a given magnet, coil, and driven diaphragm, there will be a maximum rate of change possible. If that speaker can see a signal slope greater than its maximum rate of change it will slew limit, with the same loss of information as if the amplifier or preamp was slewing. Consider than this implies the system should be bandwidth limited at both high and low frequencies to prevent signals outside the linear range of the speaker system from driving it into overload. Magnetic field irregularities may be important. The magnet structure, unless carefully designed can have eddy fields and aberrations that may adversely affect the performance.

EXTERNAL INFLUENCES. No matter how well the system is executed, the designer cannot prevent you from screwing the sonic quality up by improper use. Major problems can be caused by:

> Placing a "bookshelf" speaker in a bookshelf. Note that the shelves above and below the speaker actually form a "horn" throat. This will cause the speaker to "honk" no matter how linear its design. A bookshelf is one of the worst possible places to put a speaker.
>
> Placing a speaker in a corner. Except for specifically designed "corner horns" (and we have our doubts about those) the bass response of a speaker located in a corner will be non-linearly reinforced producing non-musical peaks and dips.
>
> Placing a speaker flat against a wall. This too will cause non-linear bass peaks and dips. Even placing the speaker near the wall can cause strange cavity resonances in the space between the speaker and the wall. Actually, the most competent speakers have available especially

engineered stands that locate the speaker “in free air.” For serious listening, the system should be located at least a couple of feet away from any wall and on its special stand if available.

Having a hard, reflective wall behind the speaker. Walls close to the speaker system will non-linearly reflect mid and high frequencies in a most obnoxious way, contributing to a “boxy” or “harsh – bright” sound quality. They will also screw up your sense of musical imaging. The wall behind the speakers should not reflect sound. You can help your sound quality a lot by draping this wall, or covering it with cork panels, treat it with upholstery fabric, or even an inexpensive foam backed carpet material. There are expensive special purpose materials available, such as Sonex, but you need not go to that extreme for satisfactory results. In general anything but hard plaster, sheetrock, concrete block, or glass is acceptable. If your room is hard and reflective, no system, no matter how expensive, will work well for you. Note, if you cannot “tame” the whole wall, you may find acoustic treatment of only the corners will be very helpful. Come out 3 – 4 feet from the corners with sound deadening material and the “box” sound may go away. General rule of thumb, if, when you speak in the room, you can “hear yourself” and you “sound boxy” then that room will be terrible for an audio system. Save your money and spend it on something else, or treat the room.

Locating big woofers in a “flexible” room. The walls and floors must be stiff for decent deep bass reproduction. If your floor is a bit like a “trampoline” then the bass frequencies will actually cause your room to work like a big balloon and will make the bass “boom.” You will also get lots of acoustic feedback into your turntable and other equipment. If you have a “balloon room” don’t try for super deep bass response, save your money, you cannot get it. Settle instead for a quality smaller speaker system, such as the Celestion SL-6 or B&W DM12 and be happy with good bass reproduction but with less power and level at lower frequencies.

Another implication of flexible floors and walls is flexible panels near your speaker systems, which is exactly the situation you may encounter in a hi-fi shop when attempting to A-B conventional speakers with flexible panel speaker systems, such as electrostatics and Magneplanars. Isn’t it obvious that the output of a conventional “box” speaker will be badly screwed up if it is placed near a big flexible panel? This is very handy for the planar speaker manufacturer, as in the A-B process they have the distinct advantage of screwing up their competition just by being there. Many planar speakers are sold because the competition was weighted in their favor as the normal system was screwed up by the acoustic coupling to the large flexible panel sitting next to it. If you want a fair comparison, listen to the planar system, and then ask the dealer to move it out of the listening room before listening to the conventional speakers. If the dealer won’t do it, well, there is more than one hi-fi dealer in the world.

The driving amplifier. It is almost impossible for the amateur to separate the “sound” of a speaker system from the “sound” of the power amplifier driving it. Thus, obviously, it is very important that in evaluating loudspeakers, you use the same power amplifier for each evaluation. It matters not that speaker A sounds wonderful on amplifier X and that speaker B sounds bad on amplifier Y if you do not know exactly how amps X and Y are reacting to their speaker loads. Which amplifier should you use?

If you are absolutely sure you are going to keep the amp you now own long term, then of course use it. Understand that you may only be “matching non-linearities” in making your final speaker selection (finding a speaker that doesn’t sound bad on your amplifier if your amp happens to be harsh sounding). But, if you plan on keeping your amp, this is a rational strategy as a high resolution speaker that sounds harsh in your particular system isn’t a good choice for you.

If you are planning for a new amplifier, use it for evaluating speakers. A word of caution, if several different speakers that have a reputation for high resolution and smooth sound seem to be harsh, boomy, or non-musical on your evaluation amplifier, you can be pretty sure that amplifier is not worth owning. A stable and linear amplifier will give good results on a wide variety of speaker designs. We do strongly suggest you try the amplifier on an electrostatic speaker system. This gives it a real tough load and will emphasize any internal slewing and harshness problems with the amp. If the amp makes the electrostatic sound smooth, detailed, and musical (never harsh or bright) then it will probably work well with any speaker.

If the dealer will not let you use your amplifier to evaluate his speakers, find another dealer. Note, of course, that you cannot expect the dealer to plug in an alien amplifier on his expensive speakers without testing it first. You should be willing to let the dealer test it to make sure it is stable, and has no DC output so that he can be sure it will not damage his speakers. Note also that if the dealer does not have the equipment or knowledge to test your amplifier for stability, then he probably doesn’t know much about hi-fi and again, find another dealer to do business with. A competent dealer should be helpful, and if he knows his business, will also be able to point out to you any sonic problems with the amplifier that are influencing your evaluation of his speakers. Listen to him, and listen to the speakers on his amplifiers too. And finally make your own decision on what combination sounds best to you.

Next month we will conclude loudspeaker evaluation by discussing how to listen, and what to listen for, the subjective evaluation process. We will also discuss Digital Audio Disc Players. A preliminary caution, don’t be the first on your block to run out and buy one. Although the sound is superb, we suspect the reliability of initial hardware and software will not be (and nobody knows how to fix them yet!), and the first generation machines are probably overpriced.

Frank Van Alstine

VOLUME TWO NUMBER FIVE MAY, 1983

I had promised to discuss Digital Audio Disc Players this month, but have run into a bit of a problem – no Audio Disc Player to discuss.

Last February, I wrote letters to the sales managers of twenty-five different Japanese mid-fi companies and Philips – all of the companies claiming to be producing a digital disc player. I asked each of them to please have their factory representative call on me with a sample machine. I pointed out to them that since we ordered equipment often from our suppliers and always paid our bills promptly, ignored promotions and selected only equipment that worked well long term, their factory reps didn’t call on us at all.

I pointed out that while we would not do the volume of a Tech Hi-Fi or Atlantis Hi-Fi, I thought perhaps one of these large suppliers might realize that it was more profitable for them to sell small amounts of equipment regularly to someone that always promptly paid for it than to sell boxcar loads to giant hi-fi marketing chains and then attempt to collect partial payment years later in Chapter 11 bankruptcy proceedings.

I was wrong. Evidently these mid-fi giants get their kicks sending their lawyers to bankruptcy court, because not a single rep has called on us with a sample disc player. I did hear from a few companies.

Yamaha told us to, essentially, "go to hell." We would have to carry every single piece in their entire mid-fi line to be able to sell digital disc players. I called the Magnavox representative (who acknowledged receipt of a copy of my letter but had not called me) and he told me "forget it," Magnavox was going to sell only to those great supporters of audio, you know, those outlets with the competent, friendly, salespeople – department stores. The Technics representative (who I know quite well from buying Dynaco through him for years) told me to forget the Technics equipment – his first order in this territory was for 10,000 units to Target (a discount general purpose chain store system similar to K-Mart) and they would be selling for $5.00 over cost. He also handles Sanyo, and told me their units were not available yet but their cost would be lower than anyone else's player. I asked him what the cost would be and he told me that Sanyo did not know what the cost would be yet. They were waiting for everyone else to announce their prices and then they would announce theirs, at a lower price, whatever that would have to be to be lowest. Although I specifically explained to the mid-fi companies in my letter to them that we were only interested in a few quality products, the Hitachi rep called me telling me he had been informed I wanted to place an order for the full Hitachi audio line, "one-brand" rack mount department store systems and all. Doesn't anyone there know how to read?

I received only one preliminary rational response, from the sales manager of Cybernet International. They actually thanked me for my interest and gave me the name of their local representative and said he would call on me. The local rep called and told me they would have a sample soon and he would get back to me when it was available. The national sales manager even answered a follow-up letter I wrote him explaining the engineering reasons behind some of their sales claims. There were solid objective engineering reasons (which the sales manager understood) behind some of their claims and I thought things were looking up. However several months have gone by and Cybernet is advertising the machines as being available at local hi-fi stores, and their local representative has never informed me of the availability of even a sample evaluation machine. So, to date, not a single Digital Audio Disc Player supplier has been willing to let me evaluate a machine.

I would not feel quite so bad about this situation if I didn't know that the Sony disc player was available, for immediate delivery, at a discount price of $675.00, and Sony discs available at $19.00. Call me for the (800) phone number of the discount house selling at that price. They don't want their name published here as other Sony dealers might not like their undercutting prices so badly so soon. Anyway, because of "marketing" we are faced with the situation that no manufacturer will talk to us, yet their dealers are selling at nearly cost to end users. The audio business is a real *Alice in Wonderland* situation.

We do advise you to wait and not purchase a first generation disc player for several reasons:

1. There probably is not a single person in the U.S.A. who can repair one at this time. If it goes bung, its back to Japan for a few months.
2. Prices will come down a lot, and soon. If you have got to be "first on your block" with one, call me for the (800) number of the discount house selling Sony units for $675.00.
3. The quality of software available now is poor. According to trade magazines, the yield of discs is about 15% at the manufacturing level. 85% of the finished discs have such gross defects they must be junked. This puts strong economic pressure on the supplier to ship out and sell every piece that plays at all, even if it has large dropouts and glitches. When the manufacturer "learns" how to produce better and the yield goes up to 90% or better, they can afford to reject all but perfect discs. Why buy grot, when if you are patient, you will be able to get high quality for a lower price within a year?
4. The first generation machines are being rushed to market due to the same competitive marketing pressures. Machines are coming out without fully integrated large scale circuits, using rows and rows of kludged circuit boards instead of the simple LSI circuits that will be available within a year or two. We have seen the insides of some of these machines. They have cards stuffed so close together there is wax paper layers between them to prevent short circuits. They run very hot. They have miles of flimsy cable set into cheap edge connectors. They weigh twice what they should.

In addition, there are good reasons for machines from different suppliers to "sound different." (If all were perfect, all would sound identical).

1. Dropouts and handling of digital errors are subject to different engineering solutions. We understand, according to Bert Whyte of *Audio*, that early Sony digital tape recorders did a crash to the rails, bottoming woofers and blowing amplifiers when dropouts got too big. We can also point out this was mentioned in *Audio* only after that model had been discontinued and a newer and lower cost model was available. Whyte's warning came only two years after the fact – honest commercial journalism.
2. Although we have heard several different prototype models at trade shows (and nearly all prototypes had excellent sound quality), this does not mean production models will. We saw the original prototypes of the Magnavox digital video disc player several years ago and at the trade show the "house sample" had resolution that was just stunning – nearly like 16 mm film. We have yet to see a production player with resolution as good as a 3rd generation video tape copy – yetch! Again, Bert Whyte reported in *Audio*, after Pioneer introduced a second generation video disc player, that the second generation machine had "just wonderful" resolution, and that the first generation machine was really pretty crappy. Again, this revelation came only after the first generation machine was out of production and no longer being advertised in *Audio*. Interesting how the candor of the professional writers tracks what is being advertised in the magazines that provide their livelihood. Do not count on any commercial writer to give any negative impressions of any performance aspect of any disc player their magazine advertises until that machine is no longer in production. Then they will tell you what a sucker you were to believe them previously.

   Thus, you cannot count on any real sonic differences to be reported in the commercial press (since the "underground" press has claimed, in advance, that "digital is terrible" you cannot count on them either). Thus, until machines are so available that you can have the opportunity to evaluate several different ones side by side (or to read reports on them you can trust), you may buy a "sour" one if you rush to buy at this time.
3. Although if done to "spec," there should be no differences between machines in reading the disc, and manipulating the digital information (outside of dropout rates and handling and reliability), there is plenty of room for "sonic" error in the filtering circuits and analog output circuits. A sloppy filter (necessary to remove high frequency digital switching transients from the final audio signal) can ring and oscillate. We suspect many mid-fi manufacturers will build as bad an analog audio output circuit as they now put in their low end receivers. (Note that my correspondence with Cybernet indicates that they know these are problems to be avoided.) Thus we do not expect all machines to "sound the same" as now reported in the major magazines.

Actually, since analog and filter circuits are our specialty, we suspect this will leave us an "opening" to design much better output stages for some digital players and provide a machine that really does work well.

Car Radio "Power"

Inasmuch as this is the time of year people start thinking about "hi – fi" car radios (a contradiction in terms) I thought you might be interested in a simple engineering discussion of "power" relating to expensive after market car radios.

1. Over the years, the term "amplifier power" has come to mean the standard IHF definition of power – that continuous average power an amplifier can put out, measured with a sine wave between 20 and 20,000 Hz, into an 8 ohm load at some reasonably inaudible order of harmonic distortion (1% or less, for example), with the full power output being measured after running the unit under test at one-third power for an hour. Although this test standard is very imperfect, it does not distinguish between closed loop or open loop distortion, it does not measure slewing or phase distortion, it does not measure whether or not the amplifier gets too hot for long term reliable operation, it does not measure power or distortion into a loudspeaker load, it does not measure out of band distortion, stability, or the amplifier's reaction to transient signals, at least it is a standard with repeatable test conditions duplicatable by any technician using good scientific methods. The standard also reports power per channel, not the power of a whole bunch of dinky channels added up, which isn't the same thing at all.

   So although imperfect, IHF does provide us with a standard, a benchmark of comparison on a crude basis between products, and to a crude extent lets the untechnical consumer make a "power" comparison for the dollar he spends on audio gear.

2. The relationship between power, voltage, and current is well known scientifically. Simply, power (in watts) equals the voltage swing of the output squared, divided by the load resistance. $P = V^2/R$. In the IHF measurement the voltage is the root mean square voltage which is the peak voltage reading times .707 , and R = 8, the standard 8 ohm test load resistor.

   Thus a standard 100 watt per channel amplifier can be easily defined as an amp that will swing 40 volts + and - into an 8 ohm load continuously with minimal distortion. Note that 40 peak volts x .707 = 28.28 RMS volts. This value squared = 800, and this value divided by 8 ohms finally equals 100 watts.

3. Now, lets note a few ways this power figure can be "fudged" by deviating from standard IHF test conditions.

   A. Lets change the load to 4 ohms, just for fun. Now assuming the amp under test can still swing the same 40 volts + and - (a quality amp will, a bad one may blow up) now our power formula will be $(.707 \text{ x } 40)^2/4 = 200$ watts! The important thing to note is that we did not change the amplifier at all, just the condition of the test. Thus the 200 watt rating is pure fiction, as it is not in accordance with IHF test conditions. The amp is still the same 100 watt per channel amp as before.

   B. Remembering that the IHF test specifies distortion, let's ignore that and push our amplifier up to 50 volts output, at which point the distortion has gone over 20% (for example). Now our power rating is $(.707 \text{ X } 50)^2/4$ (we are still cheating by using a non-standard 4 ohm load) = 300+ watts per channel!

      Our "power" rating is now three times as high, with no changes made to the amplifier. This will surely impress the prospective customer, but not the consumer fraud authorities.

      Got the point? We can tamper with the test conditions to get nearly any power rating we want, as long as we do not specify the test conditions.

Now lets look at car radio "power."

1. A car power supply is + 12 volts DC (the voltage of your car battery). That is all there is, period.

2. A power amplifier, if 100% efficient, run from a + 12 volt supply could, at the very most, swing + and - six volts into 8 ohms. This yields, at the very most possible, a grand total of two watts of car radio power.

   MOST CAR RADIO POWER RATINGS OVER TWO WATTS ARE BULLSHIT, FRAUD, AND LIES. The ratings are pure "marketing." Our definition of marketing is: Using fraud and deception to sell shit to nerds.

3. Some will claim that higher car power can be achieved by "stepping up" the car's 12 volt DC supply to a higher voltage with a converter. These same people conveniently forget to tell you that as you step up the voltage, you also (according to the rules of physics) must step up the current draw in proportion.

   Lets go back to our "honest" 100 watt per channel amplifier. Lets make one more calculation, the current required for an honest 100 watt channel. Current equals the voltage divided by the load resistance. Thus in our 100 watt amplifier with a peak voltage swing of 40 volts into an 8 ohm load, the peak current demanded of the output circuits is 40/8 = 5 amperes.

   But our car battery isn't capable of 40 volts + & -, only 6 volts + & -. Thus we must step up our voltage times 7 to get + & - 40 volts available. Unfortunately, if everything was 100% efficient we would now draw at least 7 x 5 amperes, or 35 amperes from our battery. Obviously, everything isn't 100% efficient. If we assume 90% efficiency in the converter and 50% efficiency in the car radio's output circuits, then the car battery current required is 35 amps x 1/.9 x 1/.5 = 78 amperes of current draw from your battery. If you want stereo operation, two 100 watt channels, then double the current again, to more than 150 amperes. You say you have a 4 speaker set-up and have four 100 watt channels. Congratulations, you have finally found a good use for Monster Cable (to wire direct from your battery to your car radio amplifier) for now the current from your battery is over 300 amperes for four honest 100 watt channels. Inasmuch as this is about the same current draw your starter pulls on a warm day, we suspect your radio isn't going to run for very long. You might find you are replacing alternators pretty often, and we suggest you keep a fire extinguisher handy when you use your car stereo system.

What you say, you just blew $2000 (or $4000 or even $10,000) on a "hi–fi" 100 or 200 watt car radio? Congratulations, refer back to our definition of "marketing," for you have been "marketed."

Lets look at the reality of the situation. First of all, available power even with step-up converters is in the 2 – 10 watt per channel range, and typically half of that into 8 ohms (car radio manufacturers rate their equipment into 4 ohms to dishonestly inflate their ratings).

Second, the interior of an automobile is a terrible sound environment. With the motor running, the ambient noise level is about 70 dB. Driving down the road, the noise of the road, tires, wind, and chassis resonances are added to this already unacceptable level. A large portion of the interior surface near the speakers is glass, an unacceptably reflective material for good sonic quality. The passenger volume is small and cannot support deep bass (below about 200 Hz, for example, the car interior is like being inside an underdamped boom-box speaker system).

So your automobile "hi-fi specialist" is telling you he can sell you "wonderful" sound with fictitiously rated amplifiers for use in a half glass boom-box attached to a noisy gasoline engine. Ha, ha, ho, ho, he, he! Why don't you just try parking your car in your living room, turning on the engine, and see if you like your home music system at that background noise level.

I have heard the very best the "auto hi-fi" manufacturers have to offer under the very best of their conditions, in special exhibits at major trade shows. There the manufacturers install their very best prototype systems into huge limousines. They are run on the show floor with the motors off, and with the best program material they can obtain.

The results range from god-awful to gastly! Bass that sounds like you are inside a 55 gallon oil can with someone kicking it, mids and highs that cannot be described due to the distortion from the amplifier clipping at the absurd sound levels the manufacturers insist on using in their demos. I gather a required qualification to be a representative for a car stereo company is to be stone deaf. All I can say is the sound of a $2000 car system is the same as a $50 system, only 20 dB louder – which I translate as being much worse. I might add that spending extra money on "noise reduction systems" for car use is rather futile, since the noise level of the automobile overwhelms any noise from the system itself. One use for Dolby – if you happen to own a home deck – since the playback response of the car system is terrible, record tapes for car use with Dolby on, the cruddy high frequency response of the car system will roll off the boosted Dolby highs all by itself and give you a bit more usable highs (assuming the whole thing doesn't go into overload and distortion from the boost).

It is possible to get acceptable (non-aggravating) automobile sound. A Panasonic CQ series in-dash radio cassette player (there are many similar but mine does at least continue to work) with four rational and rugged speakers, carefully mounted, will provide car sound that is not unpleasant. Spending more money only makes the system play louder, with the sound quality of a ghetto blaster. Sure there are mid-fi hi-fi manufacturers that claim their car stereo systems are just as good as their home entertainment systems. They are telling the truth, their home systems are terrible too. Put your money into season tickets to the theater or orchestra, or a side of prime beef, not into automobile "hi-fi" for there isn't such a thing.

Next month, how to listen.

Frank Van Alstine

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Usually, I would report on the Consumer Electronics Show in this issue, but I did not attend this year. If any of you did go to the show and would like to write up a report, send it to me and I will print it here if it is adequate. I can report that the most interesting things shown were probably many more digital audio disc players and the Sony Hi-Fi Beta video recorder. I understand that AR introduced a new turntable that might be worth looking into.

The new Sony Hi-Fi Beta video recorder is worth being mentioned as it essentially obsoletes all audio tape recorders (both cassette and open reel) now on the market. It can be considered as a stereo audio tape recorder that also happens to be a fine video recorder. What has happened is that Sony has put two audio tracks on the high speed video head, giving audio record and playback with 80 dB of dynamic range, signal to noise ratio and separation, with flat frequency response and very low distortion. The record and playback quality is better than that possible with a $50,000.00 studio conventional analog tape recorder. The only difficulty is that conventional "cut and splice" editing is not possible – you will need two of the machines to edit effectively. The important thing is that finally you can acquire a tape recorder that can make copies of your records (or even digital audio discs) that are essentially indistinguishable from the original. The other important thing is that the cost of the machine is about $1000.00, much less than the cost of a "high end" and poor quality cassette deck, and as a bonus, it records and plays back video too. However, you don't need to use it for that purpose. It has stereo audio inputs, outputs, and level controls (just like a conventional cassette deck) and you just hook it up to your hi-fi system like any other tape deck, only you record on Beta format video cassettes. It is not a digital system and does not have that cost or complexity. It is a very good idea. Your strategy should be to have your friends buy digital audio disc players and discs, you buy the Sony Hi-Fi Beta recorder, and make copies of their digital discs. I also understand Sony is coming out with a large library of pre-recorded hi-fi stereo video cassettes for the machine (movies, etc.) and with it hooked up to both your television and stereo system, you will probably get better sound than at the movie theater. Hi-fi lives, and things are going to get interesting in the future.

I made an error in last month's *Audio Basics* regarding being able to give you a toll free number for a source of the Sony disc player at $675.00. The toll free number is only valid for states adjacent to the discount house (which probably is not near you). I do have their normal long distance number and will give that out upon request. Again, if you have got to be "first on your block" with a DAD, why pay more than $675.00? Actually most of you missed the point. If the machines are being discounted this much already while the demand is high and supply is limited, think how fast the price will come down when the "pipeline" is filled. It is a mistake to buy one now, you will pay too much and probably get an experiment.

I have read several "underground" reports on digital audio players recently with much arm waving claiming that "digital is terrible" and it is destroying music. There were also claims that the digital players overloaded the line sections of the latest and greatest "audiophiles delight" $3000 vacuum tube preamplifiers. Interesting claims, but not well thought out. Lets take the second claim first. If the signal from a digital player, which is within the voltage specification for a line source, "overloads" an esoteric "designed by ear" preamp, we suggest the preamplifier is inadequate, not the digital player. We have no argument with the golden ear who "hears" overload, we only suggest he is, as usual, mistaking cause and effect. If his ***** class preamp overloads, we suggest it isn't a ***** class preamp, and isn't wonderful at all. We suggest it is trash. How about trying a preamp that is engineered not to overload on dynamic material instead of blaming the disc player? It is not difficult to engineer a preamp that cannot overload on any transient within its voltage rating. It is difficult to design said circuits without engineering knowledge.

As for claim number one, certainly some "digital" material will "sound bad." You are going to clearly hear; bad microphones, bad mike placement, bad mixers, bad engineering, bad musicians, bad recording environments, and everything else that is now, to a certain extent, masked by the even worse analog tape recorder and the crude mechanical record cutting and playback system. If the digital disc producer tries to sell you trash, you are supposed to be able to hear that he has tried to sell you trash. What do you want, a system that "lies" to you and tells you bad engineering is O.K.? We also agree that on some esoteric systems any digital material will sound "bad." This is simply because the system is hopelessly inadequate and cannot handle the dynamic range and transient information of digital material without "falling apart." We suggest that when golden ear reviewer claims "digital is terrible," what he unknowingly actually means is, "my wonderful ***** class playback system is terrible." Interestingly enough, well recorded digital material sounds just fine on our electronics, and we did take prototype samples of our Transcendence 400 amplifier and Transcendence preamplifier to Soundstream in Salt Lake City for actual use in their studio, on their digital masters, driving Infinity 4.5 speakers before we put them into production, to insure our electronic theory did work in practice. It did.

Finally, we absolutely agree that digital is not perfect. We agree that the anti-aliasing filters ring, that at very low output some signal is lost or changed, that dropouts occur, that some phase shift occurs, and that some extraneous "dithering" noise may be added. The point all of the Luddites miss is that the sum total of all of these problems is orders of magnitude less than the gross problems with the mechanical record cutter and playback system. Digital isn't perfect, but it is good enough now to be far from the worst remaining problem in a high fidelity playback system. The present technology is putting 5 billion bits of information on one 4" disc for an hour of playback. This is a very impressive engineering job. Consider that you would need 2500 floppy computer discs to store this much information. What do the Luddites want? The digital system can be the best musical source now. I, for one, am not willing to wait another 5 to 10 years for a "better" system and let another ten years of recorded music be ruined by the crude and distorted analog recording, cutting, and playback process.

Actually, while some claim that "digital" will "ruin" music, some very talented people (including Dr. Tom Stockham of Soundstream) are using current digital technology to retrieve "lost" music from the past.

How would you like to hear some of the great musicians of the first part of the century perform completely undistorted, in spite of the fact they were only recorded on old Edison cylinder records and 78's?

To simplify, all that is required is a sample of the recording (more than one sample of the same recording preferred), a sample of the machine used to make the recording, a sample of the playback machine, a computer, and lots of talent and patience. One can then analyze the recording machine, identify its distortion, digitize it and store it in the computer. One can also accurately separate the signal from the noise on the recording. One can then tell the computer to remove the noise from the recording, remove the recording distortion, and fill in the missing frequencies. The end result can be a "perfect" recreated recording. In fact, if one can analyze the microphone used and has knowledge of the recording hall acoustics, one can "retrieve" the sound field too, in stereo, even if the original was mono. If one gets a bit carried away, one can use the computer to create a "perfect" recording of the artist playing perfectly on a perfect instrument in a perfect hall – better than live in fact – and then one starts to wonder. Again, this is an oversimplification of the process, but save those old records and tapes folks, digital is not going to "ruin" music, but will, eventually, bring lots of great old music back to life. The amount of computer computation to do this is enormous at this time, you would probably need all the output of Cray Research for several years. But talented engineers are now making a start in this process.

Finally, to further neurose the "digital is bad" folks, consider the following: 1) Music is digital – notice all those little black dots written on the staff– its either a note, or no note – and that is digital, folks. 2) Analog tape recorders are digital machines too. What do you think is on the tape? Little particles of iron oxide, that's what. They are magnetically turned on, or off. That is digital too, dear reader. 3) How about FM tuners? The multiplex circuits are digital, and always have been. They provide left and right information by switching back and forth at a 38,000 Hz sampling rate. 4) Finally, if you are sure digital is bad, why don't you try balancing your check book with a nice analog instrument – a slide rule?

I submit those that claim "digital is bad" either do not understand the situation, or have a large vested interest in obsolete analog recording equipment. If your livelihood comes from making "direct disc" recordings, digital might hurt your pocketbook a bit, might it not? Before accepting the "digital is bad" advice, research the objectivity of the advisor.

Of course there will be those that say, "I know digital sounds harsh, dry, grainy, artificial, or whatever – the problem is Van Alstine can't hear the difference." They may be correct, I have never claimed to have any special "hearing" talent, just 30 years of experience listening. I can report that I do listen to live, unamplified music every day. With some pride, I can inform you that my teenage son has been accepted this summer into the Twin City Institute for Talented Youth Jazz Band and that he plays a pretty fair trombone. I do listen to live for at least an hour a day, and can tell live from recorded, and can also tell you there is no such thing as a ***** class "super wonderful" exactly as good as live high fidelity system. Only the Luddites make such claims. One note of "live" dispels all their claims.

A short follow up on "car radio" power. One of my readers called me to relate that he had recomputed the "power" of his "80 watt" booster amplifier based upon the data given with the power booster and my engineering formulas. Sure enough, even with a step-up inverter in his system, his real power turned out to be 6 watts, not 80 watts. Run right out there folks and spend $500+ for a 6 watt car amplifier - makes real good sense doesn't it?

Now, on to another basic topic, How To Listen To and Evaluate a System.

First of all we must a few "ground rules" and dispel a few myths. Lets get the audio myths out of the way first.

MYTH #1. You cannot remember the sound of something long term.

> Observation – when good old Joe, your long lost friend whom you haven't heard from in five years calls you one evening, saying, "Hi Fred, how is it going?" What is your response? Come on, you know it. You say, "Joe! Its been a long time since I have heard from you." Even over that cruddy, distorted, narrow band telephone you can "remember" the "sound" of Joe, five years later. You cannot "remember" a "kind" of sound? Sorry, wrong! You can remember what you want to remember. If you want to remember the "sound" of any specific audio component, you can, if you pay attention, listen carefully, know what to listen for, and if you want to remember. It takes no special talent or "better" hearing ability it only takes desire and attention.

MYTH #2. All systems sound the same to me – all I need is a portable radio to get good tunes.

> Sorry, this is just a rationalization – a feeble attempt to justify your lack of interest in the reality of music. Anyone with reasonably normal hearing can tell the difference between live, unamplified music and a playback system. Anyone (who cares) can tell the difference between a good and a bad playback system. All myth #2 tells us is that you don't care, and we feel a bit sorry for you as you obviously have not been exposed to the beauty of live music as you grew up. You are missing a lot of human culture and joy. One exception – some people can create the glory of the music internally from crude external hints – perhaps from just reading the score. I suggest it is easier when the playback system is a reasonable approximation of reality. I also suggest that good costs no more than bad, that you ought to put in the effort to select good for your money, and that you ought not to rationalize myth #2 as an excuse for not putting in the mental effort to make a reasonable value judgement.

MYTH #3. Golden Ear "underground" writers "hear better" than you do. Commercial (advertiser supported) audio writers are "tin ears."

> Sorry, there is no basis for either claim. The "golden ear" writer is just using intimidation to convince you he is some kind of authority on hearing, that he can hear audio equipment in some magic way you cannot, that he can hear wires, capacitors, (and in some bizarre way the brand of oil finish on the cabinet wood) in mystic ways you cannot.
>
> That mystic is trying to sell you his magazine, his wires, his magic capacitors,

that is what he is doing. He is using the same intimidation techniques any snake oil salesman uses to sell any mystic product – be it encyclopedias, vacuum cleaners, or political positions. He claims talents you do not have, to impress you. He hopes you won't notice his total lack of engineering ability. He may even claim to be a "respected Berkeley physicist." Is he? Check it out. Our suggestion, the more words it takes to describe the "sound" of a component in an underground magazine, the less the writer knows about the component, and the more insecure he is in his internal position. The verbage makes up for knowledge. The underground "golden ear" lives in terror that somebody will prove him wrong. He caters only to those that toady up to his mystic position. His responses to those that question his authority are crude and often obscene. If you don't agree with him you are either deaf or stupid. His subjective vocabulary to describe his constantly changing opinions of mystical sonic qualities is purposefully vague. (If you cannot pin him down, you cannot tell for sure he is wrong).

Note however, that many golden ear writers are experienced listeners. Many are good a picking out non-musical flaws in equipment. The point is, this requires only desire, training, and experience, not some supernatural talent.

The "commercial" writers do hear just as well as the underground writers. The difference is that they write differently, and for a different purpose. They have several self imposed restraints which tend to prevent them from adequately describing an audio product in print.

Their income comes from the advertising that pays for the publication of their magazine. Advertisers do not support magazines giving them "bad" reviews. Thus a conflict of interest exists between the writer's duty to tell the truth, and the writer's desire to earn a living. In this day and age, far too often, the truth does not prevail.

Many are older men, introduced to audio when narrow band IHF standard test measurements were meaningful in telling the difference between a 20% distortion 1950's amplifier and a 1% distortion Dynaco St-70, for example. The IHF standards, although too crude to determine sonic differences in modern equipment, have been perpetuated by both the writers and the mid-fi manufacturers as a way of presenting "good specifications" as a selling tool. The writers have convinced themselves that IHF specifications are meaningful, they know only how to "measure" in accordance with these specifications, and to a great extent have fooled themselves into thinking that if they cannot measure a difference then they cannot hear a difference. They have no desire to hear sonic differences.

More important, however, is their attempt at professional objectivity. They argue that since they cannot measure meaningful differences using IHF specifications, they have no objective right to report that one component is "better" than another based upon their subjective listening tests. They suggest (properly) that although they can hear differences the same as you and I can, unless they can document those differences, they have no right, or basis, to claim that because they "like" component A better, that it is, in reality "better." Thus they claim that their "likes" have no place in an objective review without objective documentation.

Thus an overview of "golden ear" writers vs. "commercial" writers suggests that the golden ears, using no objective documentation, tend to translate today's "likes" into "bests" without basis. Commercial writers, using inadequate measurement standards that cannot determine "bests," tend to make no value judgements at all, even when they hear meaningful differences. Unfortunately, neither extreme approach is meaningful to the consumer who is simply trying to acquire a good system for the money.

We should probably mention *Consumer Reports* at this time. We believe they do the worst job of all at supplying consumer information about audio components, using the worst possible combination of misinformation mentioned above. They attempt to make "beSt-worst" value judgments using inadequate test standards. Thus we see insane results such as their recent rating of a small NAD receiver as "nearly worst." Note that both the golden ears and the commercial writers have been nearly unanimous in giving the NAD a very good rating (the golden ears because it "sounds good" for the money, the commercial writers because it actually in some meaningful way – good low impedance load drive, for example – measures better than its competition). *Consumer Reports*, using their own inadequate tests – with no published documentation – found the loudness switch on the NAD to be unsatisfactory, and thus down rated the unit to near the bottom of the units tested. They rate speakers based mainly on average frequency response deviations, ignoring the fact that large deviations over a narrow band (a small average deviation) produces much more obnoxious colorations than a small deviation over a large frequency range (a large average deviation). We suggest you will do better than *Consumer Reports* advice by simply picking components at random without listening to them or reading spec sheets at all.

MYTH #4. My hearing isn't good enough to enjoy a hi-fi system.

> Can you hear the difference between live music and recorded? Unless your hearing has deteriorated to the point that you cannot tell the difference between live and reproduced, you will enjoy a quality playback system better than a bad one. My father is 82 years old. His hearing is normal for a man this age (not wonderful). His high fidelity system is still one of his greatest pleasures. Do not make your own internal barriers to stand in the way of enjoying life and culture.

MYTH #5. I need wide-band hearing ability (flat to 20K Hz) to enjoy high fidelity.

> We suspect not. We suggest that the frequencies of importance are probably limited to within 40 to 10,000 Hz. We suggest that if a playback system could be done that was perfect from 40 to 10,000 Hz, and then perfectly filtered so there was no output outside this range, even the golden ears would not notice any "missing" information or lack of bass or highs.
>
> We suggest that what we perceive as "extended highs" actually is low distortion response in the 5,000 to 10,000 Hz range. We suggest that clean, deep, powerful bass is obtained by low distortion in the 100 to 40 Hz range. We suggest you take a spectrum analyzer to a live unamplified concert and see if you can get the highest frequency indicators to move at all. We suggest you start measuring records (of musical content – not test tones) to see if there is any information above 10,000 cycles recorded on them at all. We suggest you are going to be surprised with the results.

MYTH #6. I need wide-band playback equipment (flat from DC to 500,000 Hz) for good high fidelity response. 0r as the mystics claim, the harmonics and overtones of the instruments are somewhere out there beyond a bat's hearing range, mixing and beating to form the harmonics so necessary for good perception of reproduced music on my system. Besides that, I can hear from DC to 50,000 Hz so that is what my playback system must do to.

> Wrong, wrong, and wrong. First of all, refer back to our response to myth #5. Second it is impossible to build equipment that is "flat to DC." Because of real world limitations with devices, circuits, and power supplies, if you try to go flat to DC you only get gross distortion at low frequencies. You must keep the bandwidth of the equipment within the

range it can handle. (Refer back to the St-70 rebuild – those of you that followed our advice know we are correct).

It is true that sometimes in extending the high frequency bandwidth of a component, we perceive "better highs." We suggest this is an effect, not a cause. We probably have gained better linearity in the 5,000 to 10,000 Hz range as a side effect of the design and this is what we are "hearing" as better highs.

Finally, we are not interested in reproducing everything the instrument can play or even everything we can hear. This is not the goal. The best we can do is to reproduce, without further distortion everything the microphone heard. At this time, even the best recording systems cannot record the microphone without adding distortions. What we can hear, and what the instrument can play is moot, we cannot record it. We need only design to playback what is possible to record – a much simpler goal (and one overall, not adequately fulfilled yet).

MYTH #7. IHF specifications are meaningful. Or, no specifications are meaningful, "you have got to have faith, Luke."

Wrong, and wrong again. IHF narrow band specifications are not meaningful because they do not measure all of the distortion and non-linearities in a component under all conditions of real world use. They only measure part of the distortion under an idealized set of conditions. For example, a phono preamp may measure .001% THD under bench conditions but actually operate at 20% THD when driving a tape recorder input (a very difficult load). If you cannot realize that .001% distortion in your system, then the test standard is of no value.

However, one does not then throw out the baby with the bath water and claim all engineering is bad just because IHF standards are too limited. Sorry, Luke is going to hit the exhaust port better with the targeting computer than with "faith." The answer is not, "use faith only, don't measure, measurements and engineering is bad." The answer is, learn, observe, correlate, experiment, conceive, and develop a set of measurement procedures that do measure all the distortion under use conditions.

When we can measure all the distortion, then the component with the lowest distortion (down to the level humans can perceive differences in distortion) will sound best.

Next month we will cover more myths, some ground rules for learning to listen, a discussion of the kind of source material you need to make subjective value judgments easier, and we will discuss those non-linearities that make the difference between good and bad.

CRASS COMMERCIAL COMMENT.

Our MOS-FET 200B circuits for the Hafler DH-200 and DH-220 amplifiers are getting pretty popular now. Many of our Hafler rebuild owners also use the Hafler DH-101 preamplifier. Thus it was time to do some bench research on the DH-101 preamplifier to see if we could provide a more linear preamplifier in that chassis at a rational cost.

We have done it! This is the first announcement of our new SUPER-FET 101 preamplifier. We have discarded all the original Hafler preamp circuit cards, including the power supply card. We have designed a new power supply card and audio circuit cards to fit exactly into the Hafler chassis. The power supply is a shunt zener regulator with 40,000 µF of supply capacitance after regulation and with a much wider bandwidth than the original. The phono and line circuits are identical to our SUPER-FET PAT-4 and PAT-5 circuits using LF356 J-Fet linear op amps. The phono equalization is exact (no mid-bass roll off) the circuits cannot slew limit (no transient distortion) and all circuits have a damped Q (no on-off transients or ringing). The preamp will give you the same kind of dynamics, stability, and musicality as the MOS-FET 200B amplifier. The price is $150.00 to rebuild your Hafler 101 as a "straight line" preamp (no tone controls). Add $75.00 if you have to have tone controls (it takes another complete set of circuit cards to do tone controls properly). Add $10.00 for two sets of phono inputs (not recommended as the several feet of extra cable internally adds undesirable input capacitance). Available now!

Frank Van Alstine

VOLUME TWO NUMBER SEVEN JULY, 1983.

First, I should inform you that August, 1983 Audio Basics may be a little late since I am taking a vacation (first in two years) and will be wandering around the Rocky Mountains in August instead of writing. I will have a new little mos-fet amplifier with me and if you live in Colorado, Wyoming, Montana, or between here and there and would like to hear it, write or call me before August 1st and you might get a free home demonstration.

0f course our shop will remain open and Dave and Aado will be seeing that equipment gets built promptly and properly while I am gone. If they were not competent, I wouldn't go. Remember that Aado is the designer and Dave is in charge of production so they can answer any questions you may have as well (or better) than I can.

GOOD NEWS for those of you with big amplifiers! As of July 18, 1983 U.P.S. has increased its weight limits to 70 pounds! Thus we can now ship our 400 series amplifiers directly to your door via U.P.S. instead of via Greyhound or air freight as has been necessary in the past. You can now ship Dyna 400, 410, and 416 based units to us via U.P.S. at a much lower cost.

BAD NEWS regarding the Sony Hi-Fi Beta video recorder. One of my clients has been playing with one for the past several weeks and reports there is a "sync problem" when using it as a quality audio recorder. He noticed, in making records of clean pop material with a distinct bass drum beat – boom – boom – boom – that on playback, there was a secondary and underlying bass beat – boom-boom-boom-boom-boom-boom- at double the speed of the original. He has reported the problem to Sony and was told to "make sure it was connected to a video source" for proper audio sync. Sorry – that is not the way the machine is claimed to operate and is not the way it is reported to operate in the commercial audio press. Since my client cannot get ahold of a second machine to record on (his local Sony dealer only allows playback of prerecorded demo tapes on his shop demo machine) he cannot be sure if it is a problem only with his sample machine or is a design problem typical of all. The fact that Sony "hedged" and told him to use it only with a video source is a red flag that there are design problems. Thus we recommend caution in evaluating this machine and make sure you record with it before buying to insure the bass frequencies are not being doubled and distorted by bad audio sync problems.

The Northern Pacific Audio Society was unhappy with my report on car radio power. They point out that car speakers are 4 ohms and thus the advertised power must be at least double of my calculations. They can't read – or at least cannot understand my message. Again, I will point out you can fudge a power rating for "marketing" purposes any way you want to. Why not rate power into l/2 ohm, for example – 8 car radio speakers paralleled? After all, in theory, that would make your typical cruddy car radio amp look (on paper) as if it put out 1000+ watts. The fact that it would instantly blow up into that load is immaterial, car booster amps blow up all the time anyway. The point is that people have come to expect some standard meaning in amplifier power ratings – and the standard IHF rating (continuous average power into an 8 ohm load) has some meaning to most people (just as weight in pounds has an intuitive meaning to U.S. citizens while weight in stones does not). Rating power into 4 ohms is a fake way of doubling the power rating with no power improvement. It is FRAUD! Rating power assuming a 13.8 volt supply (car engine on and racing, alternator on full charge, voltage regulator cold) is just another way of boosting the power rating without improvement in the equipment. IT IS FRAUD! IT IS MARKETING! Note that I didn't say you don't have the right to like your cruddy car radio – you do have the right to like it, even love it, even think it is wonderful. Just don't try and tell me it is good! Bad taste has no bounds.

MORE BAD NEWS. B&W recently discontinued the models DM10, DM22, and DM23 which were a series of fine bookshelf sized speaker system priced from $300 to $500 a pair. They introduced two new speaker systems at the June C.E.S. show, the model DM110 (a $300/pair two way system) and the model 220 (a $500/pair three way system). My first samples came in early this week. They are BAD! The 220 sounded like a Polk RTA 12 and the 110 (while not as bad – couldn't boom as loud with only one woofer) are still unacceptable to our ears – low definition, one note bass (loud), no real highs beyond a raspy top end, definitely not a neutral and musical sound reproducer – more of a sound maker. I suspect they will sell well – they are slightly better than the typical boom – boom speaker found at K-Mart – but are obviously designed for marketing reasons rather than sonic excellence. They do not use the long famous B&W polyester tweeter – rather something that looks like a KEF reject. Avoid these models. Note that the B&W DM12, DM14, DM7/II, 802F and 801F remain in production unchanged and remain quality products. I just don't understand where their designer's ears were in releasing the 110 and 220.

HELP! Readers – please tell me what is a good, linear, neutral, and good definition speaker system in the market at under $400/pair? Audio is getting "marketed" to death! If this keeps up we may be forced to go into the speaker making business again.

For your information, years ago, I did make speaker systems with my partner at that time, Paul Jensen. Paul was a great person, crippled by arthritis that eventually killed him in 1973. It didn't stop him from building his own house, camper, and great little speakers (for the time). We called our speakers the "Sonic Eight." They were a two way large bookshelf system (about 24" x 12" x 12") size scientifically calculated to have minimum waste from a 4' x 8' sheet of 3/4" plywood. They sold for only $50.00 per pair! The "catch" was that they were unfinished. That's right, the outside was bare grade A-D plywood, splinters and all. They sold like crazy. One day Paul delivered a set to the shop in which the big ugly "grade A-D" plywood stamp ended up on the top surface of both speakers. I gently asked Paul, "Hey, don't you think we should sand that marking off?" Paul answered, correctly, "That's finishing!" We miss Paul, and have never changed the business name. His favorite comment, when seeing typical audio workmanship on equipment coming to us to rebuild was, "That looks like it was built by a retarded ape." You would have liked him.

Paul had one interesting idiosyncrasy, he spent as close to zero on automobiles as possible and had a love for 1956 and 1957 Chevies. He would buy clapped-out used ones, cheap, and drive them until they died. One day he drove up with his latest trashpile trailing a large cloud of white and black smoke. We had to jump start him to get him home that evening, and it used 3 quarts of oil in the 25 miles home. He paid his respects to that '56 and parked it permanently. Two weeks later he came in gleefully and reported someone had driven by his house, saw the dead '56 Chevy parked in front, and offered him $40.00 for it (as much as he had paid for it six months earlier). He said he got it jump started again and watched the guy drive off in a cloud of smoke. He really felt he had made a great deal! Two weeks later he came in with a really glum face. He had heard from the Chevy buyer. The guy had actually started out for California in the car! It seems like he got all the way to Denver (nearly 1000 miles) before it blew up! Paul was so sad. He had sold the car way too soon, it would have run him back and forth for another two months here, if it lasted that long. Paul Jensen, the inventor of triple ply rubber floormats to make up for rusted out Chevy floorpans. You would of liked him. He had guts. Without him there would not have been a "Jensens Stereo Shop" and I would still be designing rural electric power lines for a living.

Anyway, back to another Audio Myth:

MYTH #8. Because I like a certain type of music (rock, pop, classical, etc.) I need to pick a playback system best for that type of music.

Mostly wrong. The object of the system is to reproduce what comes in, not to make new sounds that were not on the record. Any difference between input and output has a well defined name – its called distortion (sorry Carver, DBX, etc.).

Any predominate characteristic you can hear in a speaker system is distortion. You may like it (short term) but you cannot turn it off, and you won't like it long term. If the system is perfectly linear, it will play all kinds of well recorded music with equal fidelity and satisfaction.

Note, however, that if your taste in music runs to rock (or modern dynamically recorded classical material) played loud, you may have to compromise to have a system that does not break. You have essentially two choices:

1. Spend lots of money and acquire large speaker systems that will go loud with excellent fidelity (B&W 801, KEF 105, larger Infinities, etc.) also spending lots of money to acquire a high powered clean amplifier.

2. Compromise your choice in speakers putting durability ahead of inherent linearity. It is possible to build a smaller speaker that will go loud and stay alive. However, this requires heavy voice coils, stiffer suspensions, high power handling tweeters (such as piezo type), all design executions conflicting with linearity. In general, you can have loud, or you can have music, but not both at an affordable price.

MYTH #9. Speaker systems from different cultures have different kinds of "sound" reflecting music cultural biases of that society. (Japanese speakers are designed for Japanese music – tinkly bells and wind chimes or whatever).

Wrong. Again, a linear loudspeaker will play all kinds of music equally well, within its power handling capability. Most Japanese speakers sound "strange" because they are bad speaker systems. Japan has two problems selling speakers in the United States:

1. The Japanese manufacturers are, in general, into "marketing," not building and selling high fidelity equipment. They could care less what their equipment sounds like, as long as the spec sheets, external appearance, and commercial reviews are as good as or better than their other Japanese competitors. Consider that almost all Japanese equipment is sold in chain stores where there is no place to even listen carefully before you buy. The equipment is sold to people who never listen to it at all.

2. It is very expensive to ship "air." A speaker system is mostly a big box full of air. The bulk makes shipping costs across the ocean an important cost consideration. Those quality English loudspeakers are relatively expensive, but the end user does get quality for dollars spent. The oriental manufactures are so into low cost "discount" goods they cannot afford to build a quality loudspeaker and ignore the quality market (for it is very small).

Note that historically, the concepts for more linear loudspeakers have originated in the U.S.A. and England. The Japanese forte has been more along the lines of packaging goods to give the external (and superficial) appearance of high quality (excellent execution of "marketing").

Certainly the consumer is correct in choosing goods that have superior finish and appearance, but to a certain extent has forgotten to look beneath that fine finish to see if there is anything worthwhile inside. Remember the goal of the chain steak house – sell the sizzle, not the steak. Since "marketing" has taken over the entire spectrum of consumer goods in the U.S.A., not many people look beyond the surface finish any more.

This, of course, presents a problem to the manufacturer that does build a quality product on the inside too. How, for example, can the prospective buyer tell the difference between the typical "plastic" turntable – full of resonances and vibrations – and a Harmon Kardon T-60, for example, that exhibits excellent internal engineering and a very inert structure? Both look shiny on the outside. To a great extent, marketing gets in the way of the consumer finding quality goods because it makes objective values difficult to ascertain.

MYTH #10. I need to pick a component with lots of frequency shaping controls and/or with a certain kind of "sound" to make up for the sonic characteristics of my room and/or records.

Wrong again (mostly). Lets take "room" characteristics first.

You can make a very rapid crude evaluation of any room's suitability for a music system by simply talking in that room and listening to yourself. If, when you speak, you hear a room "echo" or you sound "boxy," then that room is a lost cause for good music reproduction as it is. Any sonic output into that room will also have excess mid and high reflections, and any system, no matter how high in inherent quality, will sound like a "box."

One must also beware of small, square, or non-rigid rooms, for they will give big bass problems. These rooms will tend to resonate at low frequencies giving an unhappy bass "boom." Obviously, in a boomy room, one should choose much smaller loudspeakers that roll off cleanly before the room gets into trouble. A speaker that plays bass cleanly, but not powerfully, is much preferable to bass boom in an overdriven room. For good bass performance, the walls, ceiling, and floors must be stiff (older homes with floors like a trampoline just don't work). Rec rooms with thin paneling make their own low frequency noises. If your are forced to live with your system in this kind of room, you may find the only good use for an equalizer, to cut bass response in the frequency range where the room gets into trouble. It may be either that, or move.

Higher frequency room resonances are much easier to deal with. Remember, we want to hear only the output of the loudspeakers, we don't want to hear our room at all, only the acoustic surroundings of the original performance. Thus it is very desirable to make our room as "dead" as possible. Any modifications of the output of the system by room reflections is wrong – again the name for differences between input and output is distortion.

Obviously, you cannot buy tone controls or equalizers that will take out obnoxious hard room reflections. You can, however, damp the room to kill the reflections. There are several inexpensive things you can do that work well.

The most important part of the room to make non-reflective is the wall behind the speakers. Suggestions – inexpensive foam backed indoor-outdoor carpeting is available in many colors, patterns, and textures. You probably can find something that is an attractive decoration as well as providing excellent damping. You can bond it to a wall just like wallpaper. The first few feet out from the front room corners are important too. Actually, if you cannot do the entire wall behind the speakers, damping the corners only will help a bunch. In my sound room, the speaker wall is covered by velvet upholstery fabric mounted on 1" thick foam backed paneling. Cork paneling works well too. One foot squares are available at most general discount stores.

Heavy drapes are useful, but not as effective as soft and dense damping fabric bonded directly to the walls. Of course special acoustical materials are available, such as Sonex, but at a much higher price (to the pocketbook and the eyes).

The floor should be carpeted. Again, an easy do it yourself project with foam backed carpet from many discount shops.

If you have a room full of glass framed pictures, modern "hard" furniture, pretty hardwood floors, and lots of big glass windows you will be forced to make a choice of priorities. This may be a room you like to live in, but it is not one your audio system can live in. You ought not to make an expensive audio investment for that room, its just not going to reproduce music there.

If you are using large bi-polar speakers such as Magnepans or Acoustats, you will find the sonic quality is much better if they are placed several feet out into the room, with the wall behind them made very dead. The object is to kill the back wave from the speaker, delaying it in time and amplitude until it is not an important part of the sonic output. You will find that depth, dynamic range, and definition really improve when the back wave is not being "splattered" right back at you from a hard and close wall behind the speakers. Yes, we know one of the owners of Acoustat recommends the speakers be placed in front of a reflective wall. I know the gentleman. His hobby is piloting his own high performance twin engined airplane. He was kind enough to give me a several hour ride once. Even with earplugs, I couldn't hear clearly for several days thereafter. I suspect his placement recommendations (which will "boost" the highs of the speakers a bunch) are much affected by many hours in that airplane.

To conclude, frequency shaping controls cannot correct for room reflection problems, for the problems are dynamic spurious resonances, not simple frequency response deviations. The correction is to remove the resonances and reflections from the room, or remove the system from that room.

Record "problems" are not easily corrected either. The problems fall into several categories, none of which are helped much by home tone controls or equalization.

1. Much pop music is recorded with substantial compression and limiting of both high and low frequencies. Sorry, tone control boost cannot "get back" what was never put on the record in the first place. Tone control boost can only make what was there louder (usually worse). Get to know your record labels, avoid the companies turning out a consistently bad product.

2. Lots of bad recording engineers use "hot" microphones located far too close to the instruments. This produces records with "shelved" boosted highs, and worse, a musical perspective totally unnatural to what one experiences at a live performance. The hinge points of tone controls just are not going to be the same as the boost from a "hot" microphone, and there is nothing you can do about bad microphone placement except avoid records made by that company.

3. RIAA equalization deviations on older records require a redesigned RIAA phono playback system to provide accurate playback gain and phase response. Twiddling with tone controls will only change the sound, not make it better. To do it right, you need to know how the original record was equalized – not something easy to find out.

4. Finally, bad engineers use too many microphones, diddle with the pan pots while mixing, use cruddy mixing boards, use honky P.A. speakers for monitoring, and there are not any "un-screw up the mix" controls on your audio system. The only cure is records done with good judgement from the recording to the pressing.

Conclusion – tone controls, equalizers, and strange sounding speakers cannot make your records sound "better," only "different" and why pay extra for "different"? Clean up your room and buy quality records. This will probably do more for you than "better" equipment will.

Frank Van Alstine

VOLUME TWO NUMBER EIGHT AUGUST, 1983

I am back from a very refreshing trip through the central mountain states. One interesting observation after crossing Wyoming several times: there is space out there – horizons 40 miles away – mountain ranges visible at 100 miles – one can drive an hour without seeing another car – there are lots of stars in the sky at night – the air is clean! Why do we insist in packing ourselves like sardines into a few dense population centers? I wonder if my eastern readers really know how big and beautiful this country is? (Big enough to confirm that my Audi will cruise at 130 mph! Speed doesn't kill! If it did all the astronauts would be dead – they have gone at faster speeds than anyone else.) Anyway –

I did find one really well done audio shop on my trip, Music West in Laramie, Wyoming. It had one of the most intelligent layouts of any hi fi shop I have ever visited. It had separate sound rooms where one could actually listen to lower priced, mid-priced, and premium audio components without distractions. Its sound rooms had good sound. The best room had Sonex lined walls and was really fine. The shop had a great variety of records, tapes, and even a good assortment of digital discs. One could go there to buy anything from a car radio to a $5000 system without finding conflicts of premises or confusion. They were equipped to handle any kind of audio customer and are doing well. How unusual to find an audio dealer that understands audio and is set up to serve well that customer.

We spent some time there evaluating digital discs on my MOS-FET CONTROL AMPLIFIER in their best sound room, using Infinity Reference II series speakers. An immediate observation is that most digital discs are very poorly engineered. Those "record problems" I discussed last month are simply much more painfully obvious when released on digital discs. I strongly suspect that this is all that is wrong with the format and that "golden ear writers" are getting confused, and blaming problems in the disc engineering onto the concept of the format itself.

We did listen to some new Telarc digital disc releases, of performances I have, and am familiar with, on Telarc records at home. The Telarc engineering is simply better in every respect on the digital discs. One other point, the shop owner wasn't afraid to open new inventory and play them for customers because, "playing these won't wear them out like it does records." Thus we suggest you request to listen before you buy new digital releases. If the performance sounds hot, bright, and grainy (many will), avoid it, it is simply badly engineered.

Another observation was that high power is not required for good playback of digital material. The Infinities are inefficient, my little mos-fet amp is just 30 watts per channel, the room was very dead, and yet we heard no overload at all (and better bass than the owner thought the speakers were capable of). Why? Because advertised power isn't real world power. Advertised "watts" are based upon single frequency sine wave signals (limited to within 20 to 20,000 Hz and with an essentially distortionless input signal) and with a simple non-inductive 8 ohm resistor as a load. This is not a good model of use conditions. In the real world the amplifier must handle distorted, wide band input signals, many frequencies at a time. It must drive a complex, and dynamically changing load. We find that most amplifiers make less than 1% of their rated power when bench tested under conditions more closely duplicating use conditions. Thus we design amplifiers to make power under transient conditions into dynamic loads. This gives us no advantage in advertised "specifications" (in fact our specifications may look "worse" than average as we use no high frequency feedback at all) but it does give us real world advantages in driving real world loads – our "30 watt" will drive Infinity speakers with a digital disc input while *The Absolute Sound* finds the same source material overloads their "pet of the month" artsy craftsy designed $3000+ esoteric trash. You don't need "more power" for digital material, you simply need real power, as the overload potential is greater.

The following is a detailed follow up on the Sony Hi-Fi Beta system problems as explained by one of my reader:

"I am writing to relay my experiences with a Sony Beta Hi-Fi SL-5200 which I purchased July 1, 1983. In playing the machine, I came across an unusual problem. I noticed that on the introduction of a pop cut which contained some drums and cymbals played in a syncopated rhythm, the Beta Hi-Fi tape contained not only the beats which existed on the record, but also phantom beats which resembled someone playing double time along with the drummer. I called Sony Consumer Information Center in Park Ridge, New Jersey and spoke to Mrs. Kokoma, a representative. The following is a list of excuses, experiments, and finally an explanation which I have received from her in regards to this problem:

1. I was told that the Sony Hi-Fi is a high resolution system, and that I may be hearing more than ever before.

   This did not get too far with me as the object of a recorder is to reproduce, not enhance the program material.

2. I was told the problem is caused by not recording any video along with the audio, preventing proper sync.

   Upon checking the tape, it did indeed have video on it. I had hooked the machine up to the cable TV prior to making any recordings. The instruction manual makes no mention of needing video in for proper audio sync. The *Audio* review in the May, 1983, issue said, "the ability to record pictures as well as such high quality stereo sound is almost an extra bonus, one that you can ignore if you so choose, by not even using it for video."

3. I was told I may be experiencing reflection due to a ground loop condition.

   I disproved this idea by using a Sony MX-14 battery powered mixer to record through instead of my system. The tape still distorted.

4. The Beta Hi-fi is a dual trace system. I was told I have an exceptional ear and stereo system which is allowing me to hear the transition between the two heads that record on the tape.

5. Sony suggests an experiment. They said to try recording with a shorting pin in the video input. This will prevent any noise entering through this input and affecting the tape.

   Note that according to the instruction manual inserting any plug into the video input will disconnect the internal tuner and prevent any video from being recorded on the tape.

   Refer back to excuse #2. It didn't help anyway.

Now for the explanation. Mrs. Kokoma said she talked to the designer of the Beta Hi-Fi system. He told her that the system uses a form of dynamic noise reduction as part of the audio signal processing. If driven with an extremely dynamic signal it may be possible to hear it pumping. At last the truth! It is a design limitation as I had suspected. I am lucky enough that my dealer will refund my money, so I will not be stuck with this machine. I am still without the Super Recorder I thought I was buying in the Beta Hi-Fi. I only hope that someone learns from my experience avoiding a lot of disappointment.

As a follow-up, I finally convinced the dealer to let me record on his demo machine. The recording had the same distortion as I had experienced with my unit. Thanks for your continued interest."

[1990 note: I have been using a more recent Sony Beta Hi-Fi for several years now and it does not have the problems of the early unit

described above. It works fine for both video and audio. FVA]

Another of my readers, Joseph Kmetz, suggests an easy and inexpensive way to make a set of speaker stands similar to that used by Celestion and B&W. He says you can purchase a section of 4" PVC plastic pipe from a hardware or plumbing store. Cut it to length. Route a 4" groove in a couple of chunks of particle board (the top one the size of the base of your speaker, the bottom one a bit larger for stability) and epoxy the pipe into the bottom board. Then fill the tube with sand and epoxy on the top board. Spray paint it the color you want and you have $100 sand filled speaker stands for $10.00. He also says there is a product called "Bump-Ons" available at art stores that keep the speaker from slipping on the stand. They are small, self adhesive pieces of rubber normally used on the bottom corners of picture frames to keep them from sliding on the wall.

Douglas Mikell writes to me from Japan to report (newspaper clippings and all) that you can expect to see the price of digital audio disc players to drop a bunch, and very soon! According to *The Japan Times*, July 27, 1983 issue, Matsushita (Panasonic-Technics) has just introduced a model SL-P7 disc player at 110,000 yen, and that Sony, Hitachi, Toshiba, and Matsushita "will compete with each other to lower CD player prices below 100,000 yen." (Note that at the current exchange rate of 240 yen per U.S. dollar; 110,000 yen = $458.00 and 100,000 yen = $416.00.) Note too that this is list price we are talking about. The discount price here could be a lot less. Remember, the TI 99/4 computer started life at $1200.00 and is now $79.00.

Regarding computer prices, it is reported that the final extreme in the current marketing suicide of personal computers will soon be promoted. It is rumored that a major new promotional undertaking will soon be announced by a major "personal" computer maker tied in with, you guessed it, a major breakfast cereal manufacturer. Sure enough, you won't need money at all to buy a computer, simply breakfast food boxtops. Save those Mumble Munchie labels folks, you soon will be able to get a personal computer for what it is really worth!

Impedances

As long as we are mentioning reader feedback, I might as well take care of the request from George Mileon, who wants **a short discussion of impedances,** especially as he is using an old tube preamp and a newer solid state amplifier.

Essentially, the output impedance of a preamplifier is determined by the characteristics of the devices and circuit configuration used. A typical vacuum tube output section can be considered to be a gain stage in series with a rather large resistor (due to the nature of the small signal vacuum tube). There is, due to the nature of the beast, a large resistor in series with the output signal.

Now, when a high output impedance preamplifier is connected to a low input impedance solid state amplifier (in general due to the nature of the power amplifier input transistors and circuit constraints), a voltage divider is formed by the R in series (preamp output impedance) and the R to ground (power amp input impedance). This reduces the preamp output signal level in the ratio of the divider. For example, if the tube preamp had a 500,000 ohm output impedance and the power amp a 10,000 ohm input impedance, the preamp output signal would be dropped by a ratio of 500,000 to 10,000 = 50 to 1, or by 34 dB!

Although this example is a bit extreme, one can expect that the output signal of the typical tube preamp will be substantially attenuated when connected to a typical solid state amplifier. In the process of turning the volume control of the preamp up to make up for this, it is likely you will clip the preamp output, causing substantial distortion.

Further, although we earlier modeled the output impedance of the preamp to be a series resistor, technically the resistor need not exist as a discrete device. The output impedance is more correctly modeled as being the output impedance of the vacuum tube itself, as modified by the biasing of it, the open loop gain of its circuit, and the feedback around that circuit. Now, when we connect this to the power amplifier, in reality the voltage divider discussed above becomes part of the feedback loop around the tube's output circuit. This lowers the open loop gain of the preamp in the ratio of the voltage divider. The first symptom noticeable to the user is a substantial increase in distortion. If the low impedance load reduces the open loop gain too much, then the circuit will have no closed loop feedback and may become unstable. There may also be further reduction in output signal level.

Note that this same undesirable condition occurs when connecting the tape output of a tube preamplifier to a tape recorder. In this case, it is the phono preamplifier section that is affected and the RIAA equalization and phono frequency response can become highly nonlinear.

In general, vacuum tube preamplifiers were designed to drive high input impedance vacuum tube power amplifiers, and do not work well with solid state power amplifiers. Our SUPER-PAS preamplifier works better than most tube preamps under these conditions as it is less dependent on closed loop feedback for stability and is driven less hard at low and high frequencies where distortion problems are most apparent.

Although some may claim that a cathode follower output stage will "cure" these problems, in practice, the current capability of the cathode follower is limited by the cathode resistor in one direction, and usually the high value cathode resistor will not supply adequate current to drive the load, thus just changing the problem, not eliminating it.

As a general rule, the output impedance of the source should be lower than the input impedance of the load it is driving. In theory this will hold true with solid state equipment where normally the output impedance of preamplifiers is quite low (600 ohms typical) and the input impedance of the amplifier is higher (20,000 ohms typical).

However, in practice this is not necessarily true, and can account for distortion not measured by IHF standard tests. A couple of examples:

1. The input impedance of most power amplifiers is set by the characteristics of the amplifier's feedback loop. Under steady state or no signal conditions, the input impedance will measure as a nice constant value as specified.

   However, under dynamic conditions (driven by a music signal) nasty things happen. When the amplifier clips, or slew limits, the feedback loop goes open and the input impedance of the amplifier goes all over the place, from very low to very high impedance conditions, presenting a variable load to the preamp, loading down the preamp and changing its characteristics.

   This of course drives the preamp into clipping or slewing, causing the preamp's input impedance (again feedback determined in most cases) to "fall apart" loading down the source. Thus, it is possible that slewing in your power amplifier can cause your phono cartridge to become non-linear. Or, another way to think about it, recovery time from overload is very slow because overload in one component can be reflected back through the entire system.

   Although no other publication has ever mentioned this problem (they don't know it exists) it is very easy to bench test for it. One simply drives the amplifier with a 10,000 Hz square wave at about half power into an 8 ohm load and connects the scope probes across the input to the amplifier instead of to the output. One will usually see ringing and oscillations at the input showing that the feedback loop is failing and the input impedance is screwed up. One can also drive one channel only into a capacitive load (a tougher test) and then watch the other channel (connected only to a simple resistive load) ring and oscillate as if it too was connected to the capacitive load.

We know of many units the golden ear writers claim to be "just wonderful" that indeed exhibit very high input impedance distortion. Again the golden ear may like this distortion, just don't try and tell me it is "just wonderful" or that the product is good. It is not. It is distorted.

We design our products to have a constant resistive input impedance independent of load or drive conditions. This may be why many people seem to like them with a wide variety of other equipment. Our units are easy loads to drive and don't screw up the equipment interfaced with them. Their characteristics remain consistent.

2. A second major impedance problem is tape recorder inputs.

In the design of a tape recorder, it is necessary to keep the very high frequency bias signal (typically about 100,000 Hz.) out of the tape input and output circuits to avoid "frying" associated equipment. This is usually done with "bias traps" consisting of complex tuned circuits. Thus the input impedance of the tape recorder can be a very difficult inductive load under dynamic (music signal) conditions.

This can load down and cause distortion in the source unless it is specifically designed to drive the load. In general, it is not a good idea to have your tape outputs of your preamplifier connected to your tape recorder (open reel, cassette, VCR, or whatever) unless you are actually using it to make a tape.

Remember, in most preamps, when you switch to phono, your phono preamp circuits are connected directly to the tape outputs in parallel with the internal preamp controls and output circuits. The tape recorder will probably screw up the phono section linearity, affecting the RIAA equalization, frequency balance, and distortion. In general, vacuum tube preamplifiers (even super expensive ones) will get into the most trouble, as their phono circuits have trouble driving even a resistive load.

Some preamplifiers have "buffered tape outputs" consisting of unity gain IC stages to isolate the phono circuits from the tape outputs. Although this is a good idea, in general it does not work as the designer intended as unity gain compensated linear ICs are working in a mode most likely to cause them to go into slewing and transient overload and the complex tape recorder input impedance is likely to cause the preamp buffer to slew and distort, thus screwing up the input impedance of the buffer stage as discussed above (input impedance distortion) and reflecting the problem right back to the phono circuits anyway.

Although our SUPER-FET preamps do an adequate job of driving tape recorders, in our Transcendence Preamp we designed each stage to have "overkill" drive into strange loads by using half-ampere peak current drive hybrid current amplifiers as output sections (National LH0002CH current amplifiers). The preamp "doesn't care" what it is driving. Many people have reported to us the Transcendence Preamplifier is making much better tape recordings than they are used to getting, even with inexpensive cassette decks. This comes as no surprise to us as tape recorders do work better when driven without distortion.

Frank Van Alstine

VOLUME TWO NUMBER NINE SEPTEMBER, 1983

The information I gave you last month regarding soon to be lowered prices on Digital Disc Players has already come to pass. No sooner was our August, 1983, issue in the mail when along came a Stereo Discounters Electronics World catalogue to me. They are already advertising the Technics SL-P7 DAD with a notation, "call or write for price." We did call (their toll free number is 1-800-638-3920) and were quoted a price of $522.00 for the SL-P7 and they told us the units should be in stock by now. If you have got to have a DAD right now, I can see no good reason to pay more for one than this. We also understand that Sears, Roebuck will have a digital disc player in their Christmas catalog this month at $589.99 with a package of three classical CDs for another $10.00.

Regarding the digital discs themselves, we have had the opportunity to hear several more Telarc releases. And they are, without question, (along with some superb M&K Realtime releases) the very best mastered source material available. We suggest you limit your disc purchases to the Telarcs at this time until everyone else learns how to record. Again, all of the sonic problems reported in the underground magazines telling you that "digital is bad," can be blamed directly upon bad engineering of the original records. When done properly, such as Telarc has done, the music is simply better in every respect than the very best analog record and record player.

Another new product we have tested, used, and like a lot is the Harmon-Kardon T-60 turntable. It has a list price of $400 (with a very competent tonearm) and is widely discounted into the $300 price range. It is the best overall engineering job on a turntable we have yet seen. It has a DC servo-drive motor (no hum field), the motor is isolated from the chassis (no motor noise dumped into the chassis and platter) (Linn-Sondek could learn a bunch from H-K). It has a massive damped platter (belt drive, of course). The platter and arm are carried on a separate suspension similar to, but much more inert than, the old AR turntable. Its shock isolation is outstanding and it has damped rubber feet too. The entire structure is as inert as any turntable we have yet examined. The tone arm is rigid, and for once has correct geometry (very unusual). Our Longhorn Grado sets up exactly in the middle of the headshell's adjustment range, aligned dead on. Even the stock turntable mat supplied with the unit is damped and inert. We have been able to improve the arm's performance with a little 1000 centistroke liquid silicon injected into the horizontal arm bearing (a tricky job we will do at no extra charge for you if you send us the T-60 and purchase our Longhorn Grado cartridge for installation in the T-60) (call us for shipping instructions). In any event, the arm is better than most separates even stock, and lifts at the end of the record. We see no reason to own anything more expensive. Sorry, we don't sell the H-K T-60, a large chain store has the franchise tied up in our area (and holds out for list price!), but you can find them discounted in most market areas. If you need a new turntable, this is easily our first choice.

The October, 1983, issue of *Stereo Review* should be of interest to you because of two interesting articles. One article because of its useful information, the other because it perpetuates the tired old commercial myth that "all amps and preamps sound the same except for minor frequency response deviations."

The useful article (within limits) is called "Jargon," by Bruce Bartlett, starting on page 57 of this issue. It does demonstrate that most audio colorations and non-linearities can be explained by frequency response deviations and reverberations and it does quite accurately match the subjective terms audiophiles tend to use with the actual non-linearity that produces the coloration heard. However, there are some well known terms missing, such as "hard," "compressed," "poor imaging," "image wander," etc. These are subjective non-linearities that cannot be easily explained as due to frequency deviations only. In fact we now get into the large non-linearities that exist, but are not tested for, in commercial equipment, which brings us to the second article:

Peter W. Mitchell (president of the Boston Audio Society) writes, starting on Page 46 of the October, 1983, *Stereo Review*, an article entitled, "How to Buy an Amplifier." The article is essentially a sanction of the testing procedures and advertising practices used by those major equipment manufacturers that advertise in (and are the source of income for the employees of) *Stereo Review*.

Understand that our argument is not with what Mr. Mitchell says, but with what he does not say. (As a trial witness, you must promise to tell the truth, only the truth, and all the truth). Mr. Mitchell essentially infers that the commercial testing procedure is complete, telling you everything you need to know about the "specifications" of products, and that only minor frequency response deviations account for audible sonic differences. He essentially sanctions the *Stereo Review* editorial policy that if it tests good under their conditions, buy it if you like the color, price, shape, and feel of the buttons.

Examples of distortion and non-linearities that *Stereo Review* (and Mr. Mitchell) never look for are as follows:

INPUT IMPEDANCE DISTORTION discussed last month in detail in *Audio Basics*. Have you ever heard of any commercial reviewer looking for this distortion? (Julian Hirsh did test for this, once, several years ago, in a *Popular Electronics* review of several power amplifiers, including our original MOS-FET 150. We had visited his lab and shown the nonlinearity to him on his bench and explained to him how to detect the problem. He did mention in that comparison article that our amp looked clean in this respect, and that several other amplifiers did in fact exhibit ringing and nonlinearities at their inputs under stress conditions, but unfortunately he also commented that he didn't know what that meant and after that, he never tested for it again in any other published review of his we have seen.) He kind of inferred to us it was not a "fair" test as it was not part of IHF standards.

WE ARE NOT INTO "FAIR" TESTS, WE ARE DETERMINED T0 FIND OUT WHAT IS WRONG WITH THE AMPLIFIER CIRCUITS. "Fair" tests are those designed to make your advertiser's products look good, not to find the real, and large problems that exist if you are willing to look for them.

If your amplifier "sounds better" when used with one preamp brand than with another (or if you are concerned about "matching components" for best sound) there is a good chance that the load your equipment presents to its driving source is unstable under dynamic conditions, obviously affecting the final system linearity. This set of non-linearities is there for you to see, Mr. Mitchell, if you are willing to look for it. It don't exist, to you, if you remain in the ostrich mode of teething.

PHASE GAIN IMBALANCES. For good perceived "imaging," both channels of your audio system must have identical phase and gain response (must be the same). *Stereo Review* (and other commercial publications) only test one channel of the equipment they report on. The frequency response and distortion tests you read are of one channel only. They "assume" that the other channel is identical if it is not obviously defective. The ostrich mode of testing strikes again. Obvious examples:

1. Conventional volume and balance controls do not track accurately. The typical control has a + or - 3 dB tolerance between sections, which can cause as much as a 6 dB level difference between channels while still meeting the specifications for that stereo control.

   It is doubtful if the careful end user can match channel levels closer than 1 or 2 dB in setting levels "by ear" on program material. In fact, most users will assume that controls do track (*Stereo Review* never told him anything different) and won't even attempt to get equal output from both channels unless the difference is really obvious. This same problem holds true with tone controls, and in the source material and phono cartridge too, but the least the electronics manufacturer can do is make his equipment track accurately and give the end user a chance.

   Of course, if both channels are not the same, imaging is bung! How about showing a plot of both channels, *Stereo Review*, showing how close both channels match, and how close the volume control keeps them matched over its operation range? Don't want to do that, do you? Would embarrass many of your advertisers, wouldn't it? Or, haven't you ever thought of it? All amplifiers "sound the same"? Baloney!

2. The internal parts used in audio circuits do not match accurately from channel to channel.

   For example, in a Hafler 101 preamplifier we recently gutted and rebuilt, out of curiosity, we pulled and measured the original RIAA capacitors before ashcanning the original circuit cards. The specified value for one particular capacitor was 2700 pF, and sure enough, that was the value printed on the capacitor.

   However, the value of a part is the value that it really is – not the value printed on its outside. In the case of this Hafler 101 preamplifier (selected at random) the "2700 pF" capacitor from one channel actually measured 2450 pF, while the "2700 pF" capacitor from the other channel actually measured 2950 pF. There was a real 500 pF difference between them. These capacitors are used to set a pole point in the RIAA phono equalization in this preamp, and the phase gain response of this circuit must be accurate if the harmonics of the recorded instrument are to be reproduced in their original perspective. (Does it sound like a violin or more like a "plastic" violin?) In addition, the phase-gain response from both channels must be the same if you want to know where the violin is.

   Stereo Review testing would never find the substantial phase-gain inaccuracies in this sample preamp. Each channel, if tested independently, would still give a "close enough" RIAA response for a good rating. *Stereo Review* tests only one channel. In addition, both channels would have to be evaluated at the same time to discover that they are, in fact, not the same. A "*Stereo Review*" never does this! "All amplifiers sound the same"? Ha ha, ho ho, he he. Only if you, and your advertisers, want them to.

As an aside, please understand that we are not "picking on" Hafler as a "bad example." The Hafler does use 10% tolerance parts in this application (more precision than most manufacturers) and both "2700 pF" capacitors were, in fact, within the 10% precision specified! Perhaps we should further deviate and tell you a bit more about "part's precision."

A "precision 1% tolerance" capacitor is not a "better" part. When capacitors are made, they cannot be manufactured to an exact value. A large production run will come out, when made of closely matched materials and production processes, within a range of values in the "ballpark" desired. Only a small percentage will be nearly exactly the desired exact value. When a company orders capacitors from the supplier, he can specify the tolerance (precision) he requires. The very tight tolerance parts, although identical in electrical performance, are much more expensive. This is because of the method the capacitor maker uses to supply and price the "precision" (1% and 2% tolerance) parts.

As each capacitor is made, it is tested and sorted. From a production run there will be a small number of pieces within 1% of the exact desired value, more within 2%, substantially more within 5%, lots within 10%, and all the rest (except for defective rejects) within 20%. The laws of supply and demand are in effect, with the small yield of tight tolerance parts selling for a much higher price. 1% tolerance capacitors may be 20 times as expensive and are hardly ever used by audio manufacturers.

One "side effect" of this process is important to understand. When you order a 10% tolerance capacitor, it is, in effect, guaranteed that it will be at least 5% out of tolerance from the desired exact value. This is because all of the capacitors that can make a tighter specification have already been culled out by the manufacturer and sold at a higher price. Thus, in general, a 20% tolerance part will be "off" at least 10%, a 10% part off at least 5%, a 5% part off at least 2% and a 2% part off at least 1%. You cannot win! Thus, even if an audio supplier claims to use 2% tolerance capacitors (which are much more expensive) he still cannot get his channels exactly balanced installing 2% tolerance parts at random.

There is, however, a way to match channels, and to get exact values where desired, and to do it without great increase in production costs.

Here is how we do it in all of the audio components we produce.

In most audio applications, it is unnecessary to have exact tolerance capacitors. For example, it is not important for the 10 µF low frequency DC blocking capacitor at the input of the Transcendence 400 to measure the exact 10 µF specified. A 9 µF value would give a roll off at 0.698 Hz, an 11 µF value would roll off at 0.572 Hz, an insignificant difference (0.126 Hz). We do, however, want to keep the pole points on both channels the same, and thus we measure and pair the capacitors (left to right channel) so that both are the same. This is easy to do (kind of like the old puzzle of having 50 white socks and 50 black socks in the drawer at night and how many socks must you take out to be sure of getting a pair? – 3 socks, of course, as long as all you need is a pair, not a pair of a specific color).

By measuring and matching capacitors, left to right channel, we insure that the gain and phase response of our equipment is identical on both channels, a requirement for undistorted imaging. While other companies may not care to, or know how to do this, we can assure you that equipment with matched pole points will sound different (and in this case better) than equipment with mismatched random pole points due to production capacitor tolerances.

We suggest that any serious audio hobbyist should acquire a precision capacitor meter. We suggest the Data Precision #938 available for about $150 thru a Data Precision factory rep in many large cities. The meter has a digital readout and measures to .05% accuracy. Then start measuring and matching. Start with the capacitors in your loudspeaker crossover networks. You will be surprised at how much better a pair of loudspeakers will image when they are both the same – and if the crossover capacitors are of different values in each speaker, the speakers are not the same.

You will need a "friendly" parts store that doesn't mind you going thru their stock of given printed value capacitors to find matched pairs. In making replacements, stay with the same type and voltage ratings as the originals. Avoid "magic capacitors" advertised by the underground golden ears. These are in general obsolete large soft film capacitors designed years ago. They tend to be microphonic (act as small "reverb" generators adding concert hall effects that didn't exist in the source) and tend to be inductive, adding high frequency transient spikes (mistaken as "detail" and "clarity") to the sound.

In those applications where great precision is required, RIAA equalization capacitors, for example, we use another simple method. Capacitors installed in parallel in a circuit add in value (a 1000 pF capacitor in parallel with a 2000 pF capacitor equals a 3000 pF capacitor). We use this little known (by audio manufacturers) phenomenon to "trim" our equalization capacitors to tolerances tighter than it is possible to buy. We do this by providing PC card locations for two paralleled capacitors for each RIAA application. Then using our precision meter and the standard range of capacitor real values for the printed value, we can select two capacitors that add up to the exact value required. This also automatically insures that both channels are the same. It takes a few minutes more of labor time, but then when we claim exact RIAA equalization for our preamplifiers, we mean it, and the harmonic balance of all the instruments will be presented exactly as recorded.

Again we would suggest to Mr. Mitchell that a preamp with exact equalization and with both channels the same will sound better. Unfortunately, nearly everything evaluated by commercial magazines is not exact, so all they hear is equipment with phase-gain problems. They may not even know that it is possible to do it right if you care.

Now it can still be argued that the phase-gain imbalances herein discussed still are just minor frequency response (and phase response) deviations and they still fall into the commercial assumption that all differences are simply minor frequency response deviations. This may be true, but as long as no reviewer reports on the accuracy, channel to channel of any audio electronics, how is the end user to even realize that he can get, for his money, equipment that is tightly matched, and images better? By not reporting on this aspect, the reviewer lets the equipment manufacturer get away with sloppy parts tolerance and poorer performance. The "if we don't measure it, it doesn't exist" reports do you no favors.

In any event, there is still a wide range of transient related distortions to discuss, distortions that can be 100% at times. Some of these very large, and obnoxious distortions present an interesting contradiction to the standard IHF measurement process. The contradiction is that the equipment can measure (by IHF standards) to be very low distortion, while, at the same time, generating very high distortion internally. There actually is no contradiction, only an error in the premise that IHF standard tests can measure all significant distortion.

This error leads the editorial staff of most commercial publications into the positive feedback loop of "if I can't measure it, it doesn't exist – if it doesn't exist, I can't hear it – I don't want to hear it because I know it doesn't exist – I can't hear it (I don't want to hear it) – I can't hear it so it doesn't exist – it doesn't exist thus there is no point of trying to measure for it – if I can't measure it, it doesn't exist —." These writers will also claim when presented with solid evidence of the existence of transient related problems that they cannot hear the problem, or that there is no proof the problem exists with a music signal as a source. They rationalize that the signal slopes of music, within the nominal 20 to 20K Hz bandwidth of interest, does not contain transients steep enough to drive even slow equipment into the forms of transient related distortion they are familiar with. They may even claim, honestly, to hear no difference between "high slew rate" or much slower, amplifiers.

We suggest however, that transient limitations should be considered like "being a little bit pregnant," the writers evaluation process is limited to choosing equipment with simply more – or less transient distortion. We doubt if any have evaluated a system in which there is no transient distortion at all. It might open their ears!

The commercial writer will also point out that the distortions in phono cartridges and loudspeakers are much larger than that from the electronics and that is where the problems lie – swamping out any minor distortions from the electronics. We would suggest that we are dealing with a different class of distortions in the transducers, primarily frequency and phase deviations and spurious resonances (although a loudspeaker voice coil can saturate and slew limit too, if driven harder than its maximum rate of change).

In the electronics, however, damaging transient effects can totally erase signals, create rapid phase "flutters," add high frequency spikes, make the power supply non-effective at some frequencies, and cause associated equipment to distort. These distortions are not at all the same as, and may be much more obnoxious than, the simpler distortions of loudspeakers and cartridges.

More about transient distortion, how we measure them, and their effects next month.

Frank Van Alstine

VOLUME TWO NUMBER TEN OCTOBER, 1983

TRANSIENT DISTORTION, what it is, how to not test for it, how to test for it, what its sonic effects are, and what to do about it. This topic will take lots of space, so lets get started.

We can generally define Transient Distortion to mean all of those non-linearities associated with rapid rates of change of the signal and to the resonances produced in the equipment by rapidly changing signal conditions.

The following is a partial list of transient distortion generators that we are aware of. Note that none of these documentable non-linearities have been discussed in either the underground or commercial audio publications.

1. Coil saturation in both phono cartridges and loudspeakers. Each of these transducers at the ends of the reproduction system has a finite limitation as to maximum rate of change it can produce or generate. If the signal slope demand is greater than the maximum rate of change possible in the device, it will simply be driven to its maximum rate of change and while it is driving as hard as it can go, all small signal riding upon the predominate driving signal is erased. The device can only go "as fast as it can" and while doing this, cannot respond to fine signal modulation at all. Inasmuch as signal removal corresponds to definition, it is, in fact, possible to measure definition, and obviously, there are real and objective differences in definition between various audio units. The "golden ears" are not "hearing things." The subjective "differences" in definition they so eloquently describe really do exist. Their problem is not understanding how to objectively describe the problems. The major advertiser supported magazines choose not to measure for this type of problem at all and claim the problem does not exist. Neither approach is helpful to you.

2. Similar problems exist in loudspeaker crossover inductors and probably as a worse case in vacuum tube output transformers. There are finite slew rate limitations in every electromagnetic device in your system.

3. Maximum rate of change limitations in every semi-conductor and vacuum tube in your system. Each device has a real maximum rate of change it can produce. Again, just like the saturated coil, if the device is fed a signal in excess of its maximum rate of change, it is driven to its maximum, and no signal riding on the predominate driving signal will pass: All small signal is again erased with corresponding documentable loss of definition.

   Keep in mind that the "enemy" signal need not be in the audio range or part of the desired music at all. As long as phono cartridges mistrack, as long we live in a world full of RFI signals, as long as harmonic distortion in one component generates high frequency harmonics passed along to the next component in line, your equipment will be exposed to signal slopes far in excess of those occurring in "music."

   And, if slew limiting occurs at any frequency, even on RFI signals far outside the audio range, while that is happening, the equipment cannot reproduce any frequencies, even those well within the audio range. Thus, if your equipment accepts RFI signals, even at small amplitude, if any internal devices are driven to their maximum rate of change trying to follow signals in the megahertz or gigahertz range, the unit will quit playing audio while this happens.

   One further observation regarding RFI input. Many "audiophiles" are now busy making audio interconnection cables out of 300 ohm antenna wire. Yep, you got it, they are turning their interconnect cables into RFI antennas! Does this change the sound? You bet! Obviously dumping a lot more RFI into your amplifier and preamp inputs will change the sound. Does it make the sound better? Not unless you like lots of RFI slew limiting and related transient distortion. Many people do like this. I don't.

4. A further side effect of slew rate limiting (driving devices to their maximum rate of change) is the distortion produced when the device goes into and comes out of slewing.

   In general, we observe that vacuum tubes tend to kind of "lump" into and out of slewing, producing what could be subjectively described as mud and murk.

   Many transistors, however, behave even worse and exhibit nasty bits of very high frequency ringing and "frizz" as they are driven into and recover from slewing. We would suggest that this is a main cause of what many consider to be "solid state sound."

   Of course there is no law that says you must design equipment subject to slew rate limitations in use, at all! If you don't allow a transistor to go into slewing, it does not slew, it does not produce harsh recovery transients, it does not have "solid state sound." We will later tell you how we achieve linear performance with no slewing at all.

5. We should observe at this time that "high slew rate" is a specification thrown about in many advertising claims these days, implying that "faster is better" and that a 500 volt per microsecond slew rate, for example, is just wonderful.

   Not necessarily. The advertisers neglect to tell you that you are dealing with a ratio problem, not an absolute. Knowing the slew rate of the equipment gives you only half the necessary data you need to know to determine if that claimed slew rate is "good." You also need to know what the maximum slew rate signal that the equipment may be exposed to. A 500 volt per microsecond slew rate is worthless if the equipment will "see" a 1000 volt per microsecond signal. A one volt per week slew rate is just fine if the steepest signal slope the equipment will ever see is one volt per two weeks! Designers, advertising men, and users all seem to forget we are dealing with ratio problems, not absolutes.

An implication of this observation is that there is, in fact, no such thing as "the absolute sound" We need only to design equipment linear enough that a human being cannot perceive any differences between the playback and reality. Non-linearities in the playback system can still exist, as long as they are below the level of human perception. Actually, the designers job isn't even that difficult. We need only to design equipment who's non-linearities cannot be perceived in comparison to the non-linearities in the source material. The source material is not live, nor the absolute sound. It is a recording! We need only to "beat the microphone." It sounds easy, but it isn't, nobody is doing a perfect job of outplaying the microphone yet.

Thus "absolute sound" is an irrational standard, all we need is "good enough sound." However, I will be the first to admit that "good enough sound" has not been achieved, and I don't even know how to define it. I can suggest however that "good enough sound" will be achieved when we can objectively define all the distortion and non-linearities in a piece of equipment and reduce them to such a low level that nobody objects to them, and then build another component of the same type in which all remaining residual distortion has been pushed down another magnitude or two, and find that nobody can hear any difference. Then, we suspect, we will have defined "the good enough sound." This might even make a tricky name for a magazine, although when all designers have achieved this goal, the magazine writers will have nothing to write about. Until then, however, we will keep trying.

6. Obviously, passive components have slew rate limitations too. But again remember we are dealing with ratio problems and the proper solution may not be "high slew rate capacitors" but more simply and inexpensively, not using a given device in an application in which its slew rate can be exceeded. The old and accurate folk saying "you can't stuff ten pounds in a one pound sack" applies and the "cure" of not trying to stuff in ten pounds usually is more effective, and much less expensive, than trying to build a ten pound sack.

7. Power supplies have finite limits too. It is important to know that the power supply of an audio amplifier can be considered to be part of, and in series with, the output circuits. All current flowing through the output must first flow through the power supply feeding the output. Obviously, if the current demands from the output exceed the maximum rate of change of the power supply, then the power supply current limits (slews) and the output slews too. Again, all signal is erased.

   Series regulators can be the worst offenders at high frequencies. In the efforts to get good regulation and hum reduction, most series regulators require many transistors in series to get adequate current gain to regulate from their reference. In general, the complex circuit ends up with more than 180° of phase shift, and is, without compensation, unstable. The instability is "cured" with lots of internal compensation making the regulator circuit bog slow. In addition, bipolar regulators are fragile and require further circuits (internal or external) to prevent them from being damaged by excess current demands, overvoltage, and from thermal runaway. The protection circuits further limit the regulator's frequency response and transient capability. When the regulator circuits go into slewing, they too produce bursts of ringing and oscillations just like the signal devices. This transient trash is fed into all of the audio circuits with quite unpleasant results.

   Thus, quite often we find audio equipment in which the design has achieved a reasonable audio circuit with good bandwidth and transient capabilities, driven by a series regulator that is slow, unstable, and very slew rate limited. The designer has only looked at the steady state operation of the power supply and is happy with the idle characteristics. The regulator may work just fine at 60 Hz and under no signal conditions, it may reduce AC ripple and hum to the vanishing point.

   However, when the audio circuits see transient information, the series regulator can current limit and put out more trash than one could ever guess. The problem is easy to observe. All that is necessary is for testers to quit looking at audio equipment as a "black box" and make a few simple measurements in the proper places. For example, when looking at transient performance, it is rather simple to drive the input with a steep transient and hang a scope probe on the power supply output instead of at the audio output. Of course, when the tester finally observes the ringing and trash output of the "regulator," it is back to the drawing board, and that isn't any fun at all.

   Because series regulators essentially don't work, we do not use them in our equipment (with one exception) even though they are a cheap way of getting good static noise characteristics. We use shunt zener regulators, in which the regulation is not in series with the output and has no bandwidth limitations at all. Because the shunt regulation is "short-proof," we can then use huge amounts of capacitance after the regulation (typically 40,000 microfarads in our preamplifiers) to achieve excellent hum reduction and stability and gentle turn on and turn off characteristics. (One cannot do this with most series regulators as the on currents into large downstream capacitors look like a dead short to the series regulators and tend to blow it up). Its more expensive to use lots of low impedance capacitors, but it sure works a lot better.

   We do use a series regulated power supply in our MOS-FET 120B, due to space limitations for adequate equivalent "raw capacitance" but with a major difference from all others. We designed a power mos-fet regulator in this application. Unlike typical bipolar transistor regulators, the mos-fet has extraordinarily high current gain, and only a single power mos-fet is required, directly controlled by a reference zener. With only a single amplification device, phase shift is less than 90°, the circuit is stable open loop, and needs no compensation at all. Due to the negative temperature coefficient of the mos-fet, no thermal compensation is required either. Because the device we choose has much higher current and voltage limits than any possible circuit demands, no protection circuits are required. Thus we end up with a 2 megahertz power supply bandwidth (much greater than our circuit requirements) and have no power supply transient limitations at all. Again, the mos-fet regulator is expensive, but well worth the effort in this application.

8. Power supplies also have low frequency limitations. As the signal approaches DC, the efficiency of the power supply drops off drastically. The impedance of the power supply capacitors becomes high, and, in active series regulators, protection circuits begin limiting output. Power supply voltage and currents drop, the output of the amplifier no longer follows the input, and heavy signal modulations show up on the power supply feeds to everything, injecting signal into the circuits at the wrong places (random and unwanted feedback loops working independently of the designer's intentions). Distortion in the audio circuit skyrockets!

   Since it is impossible to build an adequate "DC" power supply, one cannot build an audio circuit that accepts a DC input signal. The "audiophile" idea that one must "go to DC" to have good bass is wrong, wrong, wrong! By "DC coupling" your circuits or by removing the input capacitors all you do is drive the unit into much higher distortion at low frequencies. Again, many audiophile tinkerers insist on screwing up equipment by removing coupling capacitors. They must really like distortion. Don't do it! The "bass" you get is not bass at all, but simply strange low frequency noises the equipment will make while overloading and distorting at low frequencies. Anyone who claims that DC coupling "sounds better" cannot tell music from noise!

   To make matters worse, as the power supply reaches its limits, it starts to become unstable. It doesn't "roll off" smoothly, but begins to ring and exhibit high Q characteristics (much like an underdamped speaker system). This injects transient oscillations throughout the frequency range, much like the action of a "reverb" unit. And, like an echo chamber at work, this can "add" sound into the mid-range similar, under some conditions, to a "warm and pleasant" sounding concert hall. This effect has "seduced" many audiophiles into thinking that DC coupling makes their mid-range sound better. All that has happened, of course, is that the audiophile happens to like the mid-range oscillations the equipment is generating. Again, one has the right to like that effect, but, it was not in the program source. It is not music. It is simply transient distortion. It is not "good." And, you cannot turn it off.

   A directly observable effect of an unstable power supply (always present in a "direct coupled input" amplifier or preamp) is large turn-on and turn-off transients, excess woofer motion, high system susceptibility to low frequency feedback, woofer failure or system overload on Telarc drums and cannons, and sonic quality that changes with predominant AC line conditions (in particular a system that seems to get dull and muddy under low voltage or summer “brownout” conditions). If your equipment goes “whap – thump” at turn-on or turnoff, if you have to go through a “ritual” of preamp on first, wait five minutes for it to become even slightly stable, then finally turn on power amp and pray, congratulations, you own noise generating equipment, not high fidelity playback equipment.

Many manufacturers charge you extra for the pleasure of operating unstable equipment. Inasmuch as their equipment is so unstable it would blow up woofers or amplifiers at turn-on, they add an output relay to keep the high amplitude garbage out of equipment downstream until the unit finally quits oscillating. Clever, they charge you for an unreliable mechanical relay or dynamic range limiting mute FET circuit (and all the associated detection and drive circuits) in addition to the normal audio circuits.

We suggest that it is not necessary to design DC unstable equipment. We suggest that when the equipment is stable, there are no on or off transients, there are no low frequency instabilities, there is no chance of damage to other equipment if power fails, acoustic feedback and rumble driven woofer motion is minimized, loudspeakers stay alive even under dynamic bass passages, fuses don’t blow, the sonic quality stays the same even under brownout conditions, and relays and protect circuits (and their cost to you) become unnecessary.

One pleasant “side effect” of stable equipment – it plays bass! It came as no surprise to us that *Sensible Sound* claimed that our Transcendence 400 has the best bass reproduction they had yet heard. The amp is simply the most stable yet designed that we are aware of.

Or, how to design an audio amplifier that plays bass and is stable in one easy lesson:

1. Limit the input bandwidth to that required for audio signal and design the rolloff characteristics to have a damped Q.
2. Design the power supply to have a greater bandwidth and to also have a damped Q.
3. You are done, and you will have an amplifier that plays bass, not one that makes bass. Why doesn’t anyone else do it?

We are not yet finished with transient induced power supply distortion. There are lots more considerations, even when you get the power supply, by itself, to be stable, low impedance, and wide band. You still have got to connect the sucker to the audio circuits! There the fun begins again. Consider until next month the following observations:

A. The power supply can be considered to be a capacitor.

B. The connection from the power supply to the audio circuit is a wire or circuit foil trace. (A wire is an inductor).

C. The load for this capacitor and inductor is darned near unknown! (The impedance characteristics of the audio circuits under dynamic drive conditions). At best, we could consider the load to be resistive but this isn’t necessarily so.

D. What do you call a circuit consisting of a capacitor in series with an inductor terminated into a somewhat resistive load?

E. You guessed it! You call it a tank circuit – a resonator! Your power supply, as connected to your audio circuits, most likely is a tuned underdamped oscillator, doing all kinds of nasty things. We will continue with this observation next month.

Frank Van Alstine

VOLUME TWO NUMBER ELEVEN NOVEMBER, 1983

Hot flash! Digital disc players are getting lower priced, fast! Stereo Discounters Electronic World catalogue now shows the Technics SL-P7 disc player at $433.00! Their toll free number is 1-800-638-3920. We have not yet had hands on experience with this model, but, if it works at all, it should work as good as any. There certainly is no point in paying $1000-1500 for a disc player.

RFI warning! It has come to our attention that many digital audio disc players may be large scale RFI radiators. We understand that the switching circuits may dump so much radio frequency garbage into your room that FM tuners and television sets may not work at all while the disc player is running. This is a bad thing. If the disc player broadcasts high frequency trash, some of this garbage will be picked up by your amp and preamp and it will tend to drive their internal circuits into slewing. This may be one of the reasons for the “hard, grainy, two dimensional” sound complaints about disc players together with neuroses about bad imaging and ambience. It may also explain why we have not noticed the problems – our electronics are essentially immune from the effects of RFI induced slewing.

In any event, we suggest the following test for a digital disc player before you purchase. Locate an FM tuner and/or TV set very close to the disc player. Start the disc player running, and then attempt to use the TV set or tuner. Tune through all the available channels and make sure there is absolutely no RFI interference or strange sounds or pictures from the TV/FM. If the digital disc player causes any interference at all, do not purchase it.

If you find that your disc player is indeed an RFI generator, we suggest you contact your nearest Federal Communications Commission office. The world does not need more RFI pollution. Note that the problem became so bad with “personal computers” that the FCC finally stepped in with regulations for shielding and maximum RFI radiation. I have contacted my FCC office, and it looks like disc players may have been overlooked completely inasmuch as they are considered to be audio devices, while actually they are computers and require the same RFI shielding as any other computer device to avoid contaminating the radio frequency environment.

All other things being equal, we suspect there will be worse problems with the first generation expensive machines which are built with board after board full of discrete logic chips and lots of ribbon cable and skuzzy multi-pin internal interconnects. As newer machines come out, using much more fully integrated large scale special purpose devices, the layout simplifies, the amount of stray internal wiring reduces, the heat dissipation reduces (many fewer devices), the reliability improves, and with any luck at all the RFI radiation will be much less of a problem. The reason that first generation machines cost so much (aside from the marketing reasons of sticking it to the ones that have to be first on their block with a new toy) is that the clunky discrete logic circuits cost a lot more to produce than simple LSI circuits, once the LSI tooling is paid for. The $1000+ machines are not better, you just simply are paying a lot more for stone age logic circuits. Our advice still is to wait until the machines are sorted out, until they are well RFI shielded, until the prices are lower (probably under $300 for a good machine next year), until the reliability is established, and until there is a lot more good source material (until most recording engineers learn to record again).

Note that *Stereo Review* and *Audio* have once again dropped the ball – they don’t test disc players for RFI interference. If they did, they might find that some machines are actually less “perfect” than others – are you listening Julian? 0f course the “absolutes” don’t test the machines for anything – they just write purple prose about how digital is ruining music. They don’t want answers, they just want to be happy in the absence of data or facts. They are lots of help, aren’t they?

Anyway, this is probably a good time to take a break from technical matters and take a condensed look at source material – records. The following is a completely subjective listing of records I like and use, both for personal enjoyment and system subjective testing. We suspect you might find a few good Christmas present ideas here, both for others and as "hints to Santa." I have several thousand records in my collection, but only a small percentage are used over and over. I am certainly going to overlook many good records so don't take my suggestions as being a definitive list, but I think many of you will find some of the following to be worthwhile additions to your library.

TELARC RECORDS. My first choice for classic recording, primarily because of the recording skill of Jack Renner and Robert Woods. They seem to be able to consistently use microphones properly and capture the spirit of the hall and music, along with the details of the performance. The fact that they use the Soundstream digital recorder is a bonus – no tape hiss, quiet surfaces, outstanding dynamic range, outstanding resolution, and low distortion. Digital alone does not make a good or bad recording, it is simply a tool to better capture (for better or worse) the talents of the recording engineer and performing artist. Many of the Telarc works are now being issued on compact digital discs with even better fidelity than the normal records. A representative sample of Telarc records I play again and again is as follows:

DG-10072 Tchaikovsky: *Marche Slav* Leonard Slatkin, St. Louis Symphony Orchestra. I especially like the Borodin work – *In the Steppes of Central Asia* – for its string tone and strong solo instruments and imaging. It is useful for sorting out loudspeaker colorations and to quickly find out if a system images.

DG-10070 Vivaldi: *The Four Seasons* Seigi Ozawa, Boston Symphony Orchestra with Joseph Silverstein, violin. A marvelous and lush chamber work with excellent string sound and balance. It will tell you right now if your system will play a violin E string (most won't) and will also show you how badly most chamber music is recorded by others. Don't play it too loudly!

DG-10058 Gershwin: *Rhapsody in Blue* and *An American in Paris* Erich Kunzel, Cincinnati Symphony Orchestra, Eugene List, piano. This is a fun record, very dynamic, very quiet, excellent image and with many solo instruments. I like it.

DG-10045 Boito: *Prologue to Mefistofele* and Verdi: *Te Deum* Robert Shaw, Atlanta Symphony Orchestra and Chorus. A powerful and hugely dynamic choral and orchestra work, complete with a lovely children's chorus too. It will test the resolution and dynamic range of your system. The choral work is musical!

DG-10055 *Iberia*. Rimsky-Korsakov: *Capriccio Espagnol*, and Debussy: *Iberia*. Eduando Mata, Dallas Symphony Orchestra. I like the *Capriccio* because the orchestra really gets into the spirit of things and sounds "Spanish" (and slightly – but pleasantly ratty) in places. It certainly isn't a sterile reading of the work. Again, the recording has excellent dynamics, imaging, and resolution with a big and realistic presentation.

DG-10066 Mahler: *Symphony No. 1 in D* (Titan). Leonard Slatkin St. Louis Symphony Orchestra. A "titan" of a work, performed powerfully! It might turn you on to Mahler.

DG-10040 *Malcolm Frager Plays Chopin*. A solo piano recording on the Bosendorfer Imperial Concert Grand. This is a supurb piano recording and the extraordinary size and power of the Bosendorfer shines. More dynamics than a rock concert and your phono cartridge had better be able to track! (Not a recording for LS3/5As).

5038 Frederick Fennell and the Cleveland Symphonic Winds playing Holst: *Suite No. 1 in E-flat, Suite No. 2 in F*. Handel: *Music for the Royal Fireworks*, and Bach: *Fantasia in G*. (Also known as "Freddy number one" by insiders at Telarc). This is the first of the famous "Telarc bass drum" records useful on most systems for system and lease breaking – woofer reconing services should be playing Telarc royalties. However, when played on systems with stable bass performance, the drums are not overpowering, just, for once, natural. The music is exciting and the performance outstanding, reminding me of the obsolete Mercury recordings, but without the hiss and compression of early tape recorders.

DG-10065 Beethoven: *Piano Concerto No. 5 in E Flat Major, Op. 73*. Rudolf Serkin, Seiji Ozawa, Boston Symphony Orchestra. An excellent balance between the solo piano and the full symphony of a work I love.

10051 Saint-Saens: *Symphony No. 3*. (Organ). Eugene Ormandy, Philadelphia Orchestra with Michael Murray. This work was recorded in the St. Francis de Sales Church in Philadelphia inasmuch as it is a bit more difficult to move a huge pipe organ to an orchestral hall than to move the orchestra to the church. Of course the acoustics are that of a huge stone building rather than of a concert hall, but that is just fine in this case as that was the intent of the composer. This recording has been panned in the underground press as being muddy and lacking in highs. I feel sorry for those who hear it that way as it means their esoteric trash isn't working well at all. It is a demanding work, requiring perfect cartridge tracking (or the brass will spit and things will go to flanders), but everything is there on the record if you can play it back. If you play it too loud, the crescendos may blow you out the back wall of your sound room much like some of those loudspeaker commercial cartoons. Have fun.

DG-10039 Stravinsky: *The Firebird* and Borodin: *Overture and Polovetsian Dances from Prince Igon*. Robert Shaw, the Atlanta Symphony Orchestra and Chorus. Robert Shaw is a master at conducting choral works and the *Borodin* work is no exception. The dynamic range and texture is exceptional but the bass drum may overload your system.

10047 Tchaikovsky: *Symphony No. Four*. Lorin Maazel, the Cleveland Orchestra. This is my favorite Telarc recording mainly because it is so difficult to play back properly and so rewarding when you finally do. It has marvelous dynamic brass, high definition plucked strings, and awesome orchestral dynamics. You need a super tracking cartridge and great gobs of clean amplifier power to handle it along with strong and clean loudspeakers. When played back well, the record simply explodes with the dynamics of a live experience. Caution, if your system doesn't hack it, you may end up picking your woofer coils out of the back wall of your sound room – don't forget to duck as they fly by.

There are many more fine Telarc Records, but we need to give equal time to another fine record producer –

MOBILE FIDELITY SOUND LABS (Original Master Recordings). Mobile Fidelity is in the business of working from the original multitrack master tapes made by others, and remixing them and then cutting and pressing new records. Inasmuch as a lot of good music is absolutely butchered in the mixdown process and issued on records made of equal parts vinyl and sand, with a little chopped up kitty litter as filler, there is certainly room for improvement in many popular recordings, and Mobile Fidelity in general makes lots of improvements. Mixing the original multitrack tape down to the final two stereo tracks using quality equipment, high fidelity monitoring equipment (rather than the typical "PA" studio equipment), mixing for high fidelity playback (rather than for car radios) and doing it with good musical judgement is Mobile Fidelity's business and they are very good at it. In addition their pressings are typically quiet, flat, and durable. We like lots of them. Obviously, there are limits as to what they can do if the original mix is limited in quality, and with some rock groups, totally lacking in talent, who cares what is done with the mix. The following are again a representative sample of fine Mobile Fidelity recordings I like and use often.

MFSL 1-049 *Kenny Rogers' Greatest Hits*. I especially like the combined blending and yet separate definition of the vocal duet, "Don't Fall in Love with a Dreamer," with Kim Carnes. Both Rogers and Carnes voices are a bit "crusty" in life, and the job here is to insure that your system doesn't add extra edge and crust. Lots of good old fashioned "goat roping" music here with good solid bass lines and solid open imaging.

MFSL 1-041 Gino Vannelli, *Powerful People*. Some very powerful electric bass here that should cause anything in your system that can resonate to resonate. Reread speaker damping issue.

MFSL 1-025 Earl Klugh, *Finger Paintings*. Excellent high frequency transients, excellent dynamics, excellent bass. A fun record.

MFSL 1-016 Joe Sample – *Rainbow Seeker*. Lots of very clean and liquid high frequency bells and tinkly things. Kind of wakes you up easy listening jazz with good dynamics and range.

MFSL 1-020 Poco – *Legend*. A well mastered rock record with much better dynamics and range than usual. I particularly like the cut "Barbados," perhaps because I have been there a couple of times (and have a bunch of audiophile clients there – but that is another story).

MFSL 1-093 *Red Nichols and the Five Pennies at Marineland.* Mobile Fidelity has done wonders with this old (1958) Capitol master tape. This record is as fine an example of "cooking" Dixieland Jazz as you are going to find – possibly because in 1958, recording engineers had not yet learned how to screw up a live recording session. (Plastic drum sets stuffed full of rugs had not yet been invented then, either).

MFSL 1-015 Emmylou Harris, *Quarter Moon in a Ten Cent Town*. Excellent vocal sounds (a wistful little southern voice) with good blending of backup vocals and instrumental sound. The cut, "Leaving Louisiana in the Broad Daylight," is especially dynamic and well mixed.

MFSL 1-043 Crystal Gayle, *We Must Believe in Magic*. Again, fine vocals and a good system sibilance test. We use the first cut, "Don't It Make My Brown Eyes Blue," to evaluate cartridges and electronics for excess sibilance and breakup. On most systems, the vocal sibilant "s" sounds will break up, but not at all if the system transient capability is really first class.

MFSL 1-006 John Klemmer, *Touch*. The very first opening high frequency transients on cut one, "Touch," are all that is necessary to separate out harsh systems – a one second pass – fail test. Either the bells will sound liquid and pure, or like breaking glass. It is either go or no go, no agonizing required. With almost every esoteric piece we have tested, it is no go. The record is pure and musical for laid back listening.

MFSL 2-024 Neil Diamond, *Hot August Night* (2 record set). A fine vocal record, done live, with fine orchestration and clarity. Learn all about porcupine pie and chicken ripple ice cream.

MFSL 1-068 Chuck Mangione, *Feels So Good*. A major improvement over the original A&M release. One of my clients, stopping by here just after visiting a digital disc player demonstration at another hi-fi shop commented, "this has better dynamics than I heard on the other guy's disc player."

MFSL 1-089 Rickie Lee Jones. Simply well mixed and wide band high definition soft rock. Fun to listen to.

MFSL 1-073 Kim Carnes, *Mistaken Identity*. Kim's "harsh" voice is very difficult to reproduce accurately. Most systems will be far too harsh, while losing the mid-range harmonics. The vocals will be really pretty if played back properly.

MFSL 1-035 Cat Stevens, *Tea for the Tillerman*. Again, the dynamic vocals and instrumentation will be far too harsh on typical hi-fi systems, but a pleasure to hear played competently.

SHEFFIELD LAB produces a series of direct disc recordings (the output of the microphones and/or mixers is fed directly into the record cutting lathe, eliminating the tape recorder from the recording chain). The advantage is that the distortion of the tape recorder is eliminated, the disadvantage is that you cannot stop, retake, dub, or edit on the cutting lathe and thus once the blank side is started any foo-foos made require starting over again. This can be expensive, especially if the master is ruined in the stamper making process. Then, you get to call up the band again and start over. Sometimes this makes for rather "sterile" performances as everyone may be more afraid of making mistakes than of getting loose and giving a good performance. The process makes for expensive records too, as production is limited. When the stamper wears out, you cannot go back to the master tape and make a new stamper. One additional problem is recording sites are limited. You either need to bring the musicians into the studio where the recording lathe is located, or take the recording lathe to the recording location, and recording lathes are not very portable. You are not going to find many "live on location" direct to disc recordings.

In spite of the technical difficulties, Sheffield has produced many sonically superior recordings. Several of the ones I like are:

Sheffield Lab 5. Dave Grusin, *Discovered Again!* Side one, track two, "Keep Your Eye on the Sparrow," is an excellent "do-it-by-ear" antiskate adjustment test – diddle with your anti-skate until the first powerful deep bass guitar note is played with the most power, least boom, and best definition. If the anti-skate is far off, this cut probably won't track at all. Musically, I like the piano on Three Cowboy Songs and the transients of Sun Song and Captain Bicardi.

Sheffield Lab 3 (SL21/SL22) Harry James, the *King James Version*. I love to play this record and show prospective clients that everything they thought was "wrong with the record," was, in fact, only wrong with their system. The cymbals are pure and detailed (not spitty), the bass lines are distinct and high definition (not vague and muddy), Harry's trumpet never overloads the system and you can hear his distinct talent and style. The imaging is just like the photographs of the session, and the dynamics are super. If your system doesn't play it this way, your system isn't working.

Sheffield Lab 2 (SL7/SLB) Thelma Houston & the Pressure Cooker, *I've Got the Music in Me*. This record has been panned by even the commercial press. Everyone claims the vocals are "over recorded" and break up. (I think even Sheffield thinks so.) They are not and they do not. Thelma never tears or breaks up if your system can handle the dynamics. I have "floored" many people playing back this record as they cannot believe hearing it reproduced with awesome dynamics and purity.

Sheffield Lab 14 (SL43/SL44) *The Sheffield Drum Record*. I do not like this one. I don't like either the quality of the drums or the performers. The drums are not recorded with real power or dynamics and the performance is boring. Ho hum.

If you want drums that "kick ass" get M&K RealTime RT-107, Flamenco Fever. The big and powerful drum set here will knock walls out and are played with enthusiasm and skill.

I should interject here that band 3, side two of Flamenco Fever is my favorite bass transient test record. An amplifier, preamp, cartridge, or speaker with underdamped bass response absolutely will not play this cut. The drum beats will run together into an awful rumbling pool of goo. We sell lots of MOS-FET 200B amps with this record as a stock DH-200 won't play the cut and our MOS-FET 200B will. No agonizing required.

Sheffield Lab 16 *Italian Pleasures*. Excellent clean classical guitar by Michael Newman, with good string quartet accompaniment. TAS claims you have to set your balance control at 9:30 for proper imaging on this record. Interesting thought, inasmuch as the control taper will be different on different brands of preamps. This advice is much like marking an "X" on the side of your boat to note where the good fishing spot is. Anyway, its a musical and pleasant record.

Sheffield Lab 13 Lincoln Mayorga and Amanda McBroom, *Growing Up in Hollywood Town*. Outstanding vocals and instrumental dynamics. The loud vocal passages can get glassy sounding on less than perfect equipment. Everyone's demo record.

Going back to M&K RealTime Records, I also like their RT-101 recording, For Duke, a selection of "big band" jazz very well done. The cut, "Cotton Tail" has a muted cornet at the beginning that is nearly impossible to track – I have yet to hear a moving coil play it without making it sound like a scratch on the record. RT-105, "Fatha," features the musical excellence of Earl "Fatha" Hines in a very realistic way. These are both direct to disc records, as are most M&K RealTime records.

My absolute favorite classical recording is the two record set of Mahler, *Symphony No. 2* performed by the Chicago Symphony Orchestra, George Solti conducting. This is a Decca Digital recording, D229D2, and the Decca engineers flew in a set of B&W 801F speakers directly from England for monitors.

AMERICAN GRAMMAPHONE has produced a bunch of beautiful instrumental works of outstanding sonic quality (their covers are lovely too). Among their records that I keep going back to are AG-365 *Fresh Aire III*, AG-368 *Daydreams*, AG-370 *Fresh Aire 4*, and AG-373 *Fresh Aire Interludes*. The last two are digital recordings with no loss in musicality at all.

Jazz lovers should be told about the North Texas State University Lab Band. This school, address North Texas Lab Bands, Box 5038, North Texas Station, Denton, Texas 76203, is one of the foremost music schools in the nation. Each year a Lab Band jazz record is cut of excellent musical quality. Big band jazz fans should write for their catalog.

A record I love, and surprise people with, is one of my oldest records, a 1958 Columbia, CS 8082, Newport 1958. Its a live recording of the original Dave Brubeck Quartet that just sounds real! Again, it was recorded before Columbia learned how to multi-track and diddle the mix. CS 8192, a later Columbia release, although nowhere near as well engineered, contains the Brubeck Quartet playing the Paul Desmond composition, "Take Five." This is a classic jazz milestone written in 5/4 time. If you are not familiar with it, get it.

NAUTILUS RECORDINGS also produces excellent premium recordings. I like the dynamics of NR2, *Aspen Gold*, a new digital recording of the Kingston Trio which brings back memories of the 1950's.

Their remaster of Fleetwood Mac, *Rumours* (NR 8) is just fine, and I use their remaster of Linda Ronstadt, *Simple Dreams*, (NR 26) to test the definition of other audio systems. It seems that during this recording session, there was a "walk on" by another female singer, a good friend of Linda's, who "sat in" on a few takes. This singer gets no credits on the album and on most systems, you will never hear her. To really test the definition of your system, simply get this record and play the cut, "I never will Marry." If it sounds like a multi-dub of Linda, you lose. If you cannot tell me who the second female singer is, you still lose. This cut puts a lot of arrogant audiophile salesmen down when they find that despite their claims, they cannot "hear" the second singer instantly for who she is.

The Japanese are getting into the remaster act too. The Toshiba-EMI series of Pro-Use recordings are outrageous. ALF-99002, Holst, *The Planets*, Previn (the original EMI recording is used by many golden ears) remaster makes the original sound like a cassette. The EMLF- 97002 Pink Floyd, *Dark Side of the Moon* makes the original sound like an 8-track, and the EALF-97001, Beatles, *Abbey Road*, makes the original sound like an Edison cylinder record. You should have a few Pro-Use Toshiba-EMI records in your collection. I also like many of the Japanese AUDIO LAB series of jazz recordings. Although they are older, they are demanding, and will not sound really musical until the system is really musical. ALJ-1042, *Side by Side 2* has interesting jazz piano cuts. The EAST WIND series of direct cut jazz recordings featuring small jazz groups also are only useful on very high quality systems. They tend to sound hard and dry at first, but are actually close and intimate if the system is really cooking. Not so the OPUS 3 jazz recordings. These have a high end "lift" that is most annoying – miked too close and hot for me.

CRYSTAL CLEAR RECORDS have produced several fine direct disc records. I like Laurindo Almeida CCS8001, Virgil Fox, The Fox Touch Volume Two, CCS7002, and San Francisco Ltd., CCS5004. In particular the FOX recording features the deepest bass notes ever recorded (sustained 26 Hz organ) which feels like (you cannot really "hear" it) the change in air pressure as a big thunderstorm passes.

Finally I should mention Robert Fulton's recordings on his ARK label. His sampler record, *The Fulton Gold Series Vol. 1*, ARK No. 4170-S is worth owning. The recording engineering is just great, putting things in space and time with unexceeded ability. The harmonic balance is absolutely natural, and the definition is just fine. There is however, a "catch." Bob records amateur "talent" (high school bands, church choirs, etc.). Thus an interesting thing happens as your system gets better. On a so-so system, the records sound pretty good. As the system gets more dynamic and focused, the records get a lot better. But finally, when the system gets to a certain level of definition, the records start sounding bad again, because now you can hear how bad the "talent" is. When each voice in the choir is individually audible (each slightly off key), when the solo female singer's "Elmer Fudd" lisp becomes obvious, then the records are not fun any more.

Frank Van Alstine

SPECIAL BONUS ISSUE PERSONAL COMPUTERS, 1983

EVERYTHING YOU DIDN'T WANT TO KNOW ABOUT PERSONAL COMPUTERS AND WERE AFRAID TO ASK.

[1990 NOTE: DEAR READERS, THIS ISSUE IS INCLUDED HEREIN BECAUSE THE DECEMBER, 1983 ISSUE (AND JANUARY, 1984) WAS A NOW OBSOLETE DYNA PAS REBUILD PROJECT. WE DON'T WANT YOU MESSING WITH THE 1983 PROJECT BECAUSE IT HAS BEEN REPLACED WITH THE CURRENT SUPER PAS THREE REBUILD. CALL US FOR DETAILS IF YOU WANT TO REBUILD A DYNA PAS NOW.]

I am writing this special issue mostly to "distill" in writing my own "adventures" into the personal computer market over the past two years. We (Jensen's Stereo Shop) need a competent computer now. We need it to do advanced circuit analysis, to organize my customer files into a rational file and mailing list, to do text editing and spelling, to allow myself, my associates, and our kids to learn "computing" as a future marketable skill, and to reduce the routine paperwork of bookkeeping and accounting. We need a machine to allow us to serve you and ourselves better.

For the past two years we have explored the market, evaluating what was available, ignoring manufacturer's claims and advertising, and testing what the machines really can do. In general, we have found that they cannot do very much.

We should start this discussion by defining what a computer should do.

1. First of all it should compute. It should mathematically manipulate numbers rapidly, with great accuracy, and with adequate precision for the intended task.
2. It should be easy to program. It should be possible to tell the machine what you want it to do in a rational high level language.
3. The machine should have adequate storage capacity (memory) to allow one to have the operating system, the program in use, the data pertaining to the program, and the results of the program quickly accessible without transferring information slowly back and forth into some secondary storage medium.
4. The machine should have useful and rapid readout of data for human use. This includes a high resolution video display that can show information in its final format without eyestrain and a printer that reliably and rapidly prints out.
5. The machine must be designed to defy "Murphy." It should be difficult to make the operating system, user program, or accessory crash, either accidently or on purpose. It must be nearly impossible to destroy files and programs. It must "fail safe" in the event of power interruptions.
6. Note that a criteria we have not included is "there are great gobs of pre-written programs available to run (maybe) on this machine." General computer salesman advice is to "evaluate the software (programs) available and then choose a machine that has the hardware to run the programs you want to buy." We will demonstrate herein that this is very bad advice, that one must first evaluate the hardware (we will attempt to here for you) and then pick a machine that is competent.

A brief history of computer development is in order at this point. The electronic computer (aside from mechanical devices with roots in the 19th century) is a child of World War Two and the Manhattan (atomic bomb) Project. The first vacuum tube computers were developed to execute simple, but time consuming, mathematical problems that could not be done fast enough on the mechanical calculators of the time. They were huge (your living room full of 12AX7 tubes), unreliable, hot, incredibly expensive (only the government could afford one) and were "hard wired." Only specially trained engineers and physicists could "program" one to solve a specific problem, by, essentially, changing the interconnections on thousands of wires to different places in the circuits. In the late 1940s slightly improved versions appeared in research labs and major institutions, again, requiring thousands of engineering hours to program, use, and maintain.

The vacuum tubes were used as electronic switches, either turned on, or off, to represent one bit per tube of digital data. As the transistor (developed by Bell Labs in the 1930s) became a practical, reliable device, it replaced vacuum tubes in the computers of the 1950s. This made the computer practical for major businesses (banking, science, universities, industry). At this time some external equipment to make programing and use more practical was developed, such as the "IBM" punch card, paper punch tapes, teletype keyboards, and magnetic core memory. However, highly skilled professionals were still required to operate the computer and to tell it what to do, and the computers were still huge, hot, and expensive, containing hundreds of thousands of individual transistors.

In the 1960s, integrated circuits were developed (essentially the packaging of many small transistors and their circuit interconnections on a single silicon base in a small multi-pin package). This allowed the "distillation" of thousands of transistor switches from occupying many square feet of circuit board space to a few square inches, and brought the size, cost, energy requirements, and heat dissipation of the computer down to a more practical level. Among the first of the "minicomputers" were those from Digital Equipment Company (DEC). They were used, essentially, as auxiliary equipment to large "mainframe" computers by research institutions such as Bell Labs, because, although they were useful for running programs (making calculations) they were much more difficult to program (tell them what to do) than mainframes, and for practical operation, the programs were written on the mainframe and then loaded into the minicomputer to run. Obviously, owning a minicomputer independently was not practical unless you also had direct access to a mainframe to program it.

One of the most important developments of the early was the development by two Bell Labs engineers, Dennis Ritchie and Ken Thompson, of a rational operating system for the DEC minicomputer. This set of instructions, named UNIX, allowed the minicomputer to be programed and used independently of the mainframe. This brought the computer to the use of mid-size commercial businesses. Now for $500,000 to $1,000,000 one could have the bookkeeping, accounting, payroll, scientific computing, and data management capability that cost many millions only a few years before. It also became possible to train nonscientific personnel to write useful programs and the occupation of Computer Programer came into existence. Obviously, in the same timeframe, other business minicomputers and operating systems came into existence. We mention DEC and UNIX as the most visible example of minicomputer development.

Since that time three "tracks" of computer development have evolved. Track one is the further improvement of the speed and capability of the mainframe, on a nearly cost is no object basis, resulting in today's "supercomputers" such as the Cray and the Control Data Star. The second track is the development of current mainframe capability into more affordable minicomputers by large scale integration of circuits and superior operating system development. We would normally have expected, over the last decade, heavy competition along the second track to bring very competent and affordable minicomputers within price reach of individuals. However, and very unfortunately, the second track of development has been to a great extent "short circuited" due to the unexpected development of a third track, the from the "bottom up" development of the microcomputer.

In the 1970s, mainframe and minicomputer development proceeded using general purpose integrated circuits — memory chips, logic chips, buffers, adder chips, shift registers, and so on. There was no general push to put all computer circuit functions on one chip (integrated circuit). However, in the early 1970s a small California electronics company, Intel, entered into a joint venture with a Japanese calculator manufacturer to develop a crude programmable calculator which required the development of special purpose integrated circuits combining a large number of general purpose computer circuits into one small package. Although the joint venture did not succeed, Intel was left with the first general purpose microprocessor chip. People at Intel then realized that general instructions could be written for this chip (Intel 4004) and would allow hardware designers to develop useful control processors using a microprocessor and memory as part of the design, rather than having to "reinvent the wheel" and design a special purpose discrete microprocessor for each engineering application.

It is important that you understand that these first "microprocessors" and the slightly more competent second generation 8-bit microprocessors now in use were intended as special purpose control processors, for applications such as controlling industrial drilling machines, industrial chemical processes, security systems, and so on. The designers never considered them as general purpose computers, for they are far too slow and limited in memory usage and mathematical capability to do useful computation.

However, an interesting situation then occurred. Clever people formed small companies such as Altair, and begin using these 8-bit microprocessors to build simple "hobbyist" grade general purpose computers. They were slow, crude, very difficult to program (throw rows of toggle switches to generate "machine code"), but, compared to what was available, very inexpensive (a few thousand dollars) and were within price reach of an individual and would fit in his home. They quickly attracted the attention of the fanatic "hobbyist" crowd but were not useable by those without special training in electronics and mathematics. Thus early microcomputers had no broad market appeal.

Then Apple changed things and changed the entire face of the computer business by introducing three things simultaneously: They packaged an 8-bit microprocessor based computer in an easy to use and attractive package, complete with keyboard, and jacks for external video displays, disc drivers, and printers. They provided an operating system and programing language that allowed use by non-technically trained end users (anyone can learn to program in BASIC). They then promoted the package as something real neat for average Joe to own and use. They succeeded beyond their wildest imagination and made personal computer a household word and a desired product by most everybody. They started a huge feedback loop of thousands of people writing programs on an Apple (because there was no other early alternative) and then thousands more buying Apples because there was so many programs available. Unfortunately, this is where, for all practical purposes, development of this third track has stopped due to the basic limitations of the 8-bit microprocessor. Remember these personal computer companies are not the microprocessor circuit designers, they simply package other people's circuits. Even the IBM personal computer is built with Intel supplied computer chips.

Since the first Apple, only small refinements have been made. Personal computers have been made smaller, more reliable, with better accessories, and at lower and lower cost (trade sources say that the Apple lle costs $200 to produce and will soon sell for $600 at retail). Similar machines, Commodore 64, etc, are available under $300, and one can learn a lot about BASIC programing with a simple Timex 1000 for under $50.00. But what is difficult to learn about the current personal computers, without experience on much more sophisticated machines, is what they will not do. Unfortunately, marketing has taken over from engineering research and advances in the third track of personal computers.

Essentially, all personal computers are based on one of three 8-bit microprocessor integrated circuits (or clones thereof): The Intel 8088 (a third generation chip that can access more memory than the typical 8-bit processor - used by IBM and the new TI personal computer and IBM clones). The Signetics 6502 (used by Apple and a great gaggle of Apple clones). The Zilog Z-80 (used by Radio Shack, Sinclair, and Commodore). There are not major differences between the function and performance of these "basic three" microprocessor chips. Some claim that the 6502 is a bit easier to write machine language commands for (nice if you are a math major), it is thought that the Z-80 runs a bit faster (but still far too slow for hard computing), and there is a bit more support (external circuits and factory support) for the 8088. The important thing to know is that programs written to operate on any one of these microprocessors cannot be directly transferred to a machine using a different brand of microprocessor due to internal differences in the chip designs and differences in overall circuit configurations.

An interesting detail to know at this time is that the microprocessor makers may occasionally change the internal construction of their processor chips. They will still interface just fine in the intended computer, and work just fine on all programs written in the programing language designed for that computer. However, sometimes "software" writers get too clever for their own good, and write programs in "machine code." By "bypassing" the normal user programing language they can write programs that run faster, take less memory, and are difficult to copy. However, most of these clever writers don't understand that when the chip maker changes internal specifications that their programs will crash and burn. In machine code, the program directs the computer to put information into and take information from specific memory locations inside the computer. When a chip is changed, that specific location may no longer be there at all! A program using that specific location will then crash! Commodore recently caused great anguish with one word processing software company. Commodore changed processor chips to fix a "color" bug in the Model 64. The word processing vendor had written their software in machine code. Every single software set they had produced crashed (thousands of floppies) (thousands of unhappy users) on the newer machines. Dealers didn't know what was wrong because the programs ran fine on their older demo machines. The software writers didn't know that Commodore had changed chips. Commodore didn't tell them because the new chip worked fine when the computer was used within its specified design limits — write programs in the high level language for that machine. The moral, machine code will only run on other computers with identical microprocessor chips, internal timing, and accessories. Don't buy programs written in machine code or write machine code programs expecting that others will be able to successfully run your programs.

Now, back to 8-bit microprocessors. An 8-bit processor can handle a string of eight digital bits (from 00000000 = 0 to 11111111 = 255 = 2^8-1). This is an adequate list of numbers (from 0 to 255) to assign a number for each letter of the alphabet, both lower and upper case, for punctuation, for the numbers 0 thru 9, and for adequate special function keys. But the computer has to have code numbers to tell it more than just what something is. It must also be told where to put that something, and what to do with it. Since the 8-bit processor can, in one cycle of operation, process only a single 8-bit "word" (one byte) obviously it takes at least three cycles of computer operation to tell the computer what something is, where to put it, and what to do with it. And that would be only to manipulate one character of data! Consider how many processing cycles it would take to just print out this page.

To be fair, we must point out that in one respect, the 8-bit processor is more than 8 bits. Computer memory can be considered to be a rack of pigeonholes such as found in old post offices. Each slot can hold one 8 bit word in an 8 bit computer. Each slot must have a unique address just as your house does, so the processor (mailman) can locate where to put information and where it is located when it needs to retrieve the information. If an 8-bit processor could only handle addresses from 0 to 255, it would not be able to address and store enough memory to write your name.

The good news is that useful 8-bit microprocessors can manage 16 bit words for locating memory addresses. This gives it the ability to "address" 2^{16} = 65,536 (commonly called 64,000 or 64K) memory locations. This would be enough memory to hold about 14 pages of typing, IF, that was all that was required to be kept in memory.

The bad news is that text is not all that is required to be kept in memory. The memory of the computer must hold first the machine code "housekeeping" instructions that for example, tells the computer how to turn on. These basic turn-on instructions are permanently located in Read Only Memory (ROM) chips in modern microcomputers. Then the operating system (the master program that provides the necessary capability to allow user program languages and programs to function) must also reside in memory. The programing language (BASIC, FORTRAN, COBAL, etc.) must reside in memory. All the commands to operate disc drives, video screens, keyboards, modems, etc. must be located in the memory. Therefore your 64K memory isn't 64K of memory available for you to write programs and hold useful data. Much of it is used up before you even start. If the operating system is brief, then the necessary instructions have to be written into every user program taking up more memory. You cannot win.

More bad news is that obviously, 64K just is not enough memory. For example, the operating system and BASIC programing language for the Hewlett Packard HP-9000 32 bit computer we are currently considering requires about 500,000 bytes of memory just to hold this master program. This causes no problems with this enriched 32 bit machine because its processor can "address" over 4,000,000,000 bytes of memory. It would obviously be impossible to use such a flexible and enriched operating system on an 8-bit computer. There isn't any place to put it.

Some 8-bit computers claim to have more than 64K of memory. Outside of the Intel 8088 processor based machines (IBM and TI) which have more memory address capabilities than normal 8-bit processors (up to 1,000,000 bytes if the computer is designed to take full advantage of the processor chip), an 8-bit processor can only directly address 64K of memory. Any "additional memory" beyond 64K is not directly and transparently accessible. It must be independently called, accessed much like an accessory disc, and runs much slower than direct access memory. User programs cannot overrun the basic remains of the original 64K of memory or cross between additional memory banks. It is not more useful than a floppy disc drive.

The real problem with an inadequate memory is that a very enriched and easy to use operating system and programing language cannot be done. There isn't room for it. This makes programing a "pain in the ass." This is the reason there are so many programs purchased for personal computers rather than being written by the end user. Writing rational programs on a personal computer is very difficult and time consuming. Thus the user is at the mercy of someone else's program which may, or may not, do what the user wants to get done. Why do you think there are hundreds of different word processing, spelling, filing, mailing, and guessing programs on the market for personal computers? It is because none are very good, really easy to use and crashproof, and complete and fast. None are good because the operating systems are too crude to allow competent program development. Consider that with a personal computer, to write a program, you must start with line #1, and proceed from there to the bitter end, keeping track of every GOTO, IF-THEN, and RETURN. If, after about 500 lines of program, you find you need another loop somewhere in the middle, you need to then remember how to re-arrange all line numbers and conditional loops so they get back to the right places, a mess getting beyond rational human capability.

Consider for a moment how easy it is to write programs on a competent computer with a competent operating system. For example, a complete spelling program can be written on a UNIX operating system on a minicomputer with the following instruction line:

lcase <ab | colm | sort | uniq | comm dict (that is all there is to it!)

lcase <ab tells the computer to take the file to be checked (in this case file AUDIO BASICS =ab) and change all the capital letters to lower case letters because there is no need to clutter up the dictionary with duplications of words just because the first letter might be capitalized if it happened to be at the beginning of a sentence.

colm tells the computer to take all the words in file ab (now without any capital letters) and put them in one long list, or column, instead of in sentences.

sort tells the computer to now sort this list into alphabetical order.

uniq tells the computer to eliminate redundant words (we need only one sample of each word in the column).

comm dict tells the computer to compare this list of words with the list of words already properly spelled and stored in the file dict (dictionary) and to print out all the words that are not in that dictionary (all the misspelled and unique words).

Obviously, a couple of more simple commands will correct all the misspelled words at the same time, and can send on to the growing dictionary all the unique, but correctly spelled words for future reference. The whole thing will run in a few seconds.

Is there anyone out there that wants to describe a spelling program for an 8-bit personal computer and the meaning of each command in half a page of printed text?

The above is an example of why you pay about $300 each for poor programs to run on a personal computer and you don't pay anything, you write your own programs, when using a competent computer.

One note about "UNIX." This Bell Lab developed operating system, fully implemented, is very competent. Bell Labs licences the UNIX system to computer manufacturers. One minor problem, subsets of UNIX have also been developed that contain only a small fraction of the complete UNIX commands and structure. Bell has not in the past distinguished between the complete UNIX system and much less capable subsets. Thus a computer company can claim to provide a UNIX based operating system without offering much that is useable at all if it is a limited subset. A complete UNIX system requires about 2,000,000 bytes of memory and obviously won't run on a personal computer.

Another problem of the 8-bit processor is that it runs slowly. Because it can handle only 8 bit words, it takes several cycles of operation to manipulate any given piece of data. With more competent 16 bit and 32 bit machines, the data, the command for where to put it, and the command for what to do with it can all be given in the same word, and the manipulation of that data may take only one cycle. In addition, 8-bit processors operate typically at 1 to 2 megaHertz, 16-bit machines at 4 to 8 megaHertz, (two to four times as fast), while recent 32-bit machines run as fast as at 18 megaHertz rates, (two to four times as fast again). To give you an idea of computer running time consider the following simple BASIC program. It will run on an HP-9000 in about 10 seconds. Try it on your computer with a value of 1,000,000 for N.

```
10 INPUT "N";N
20 LET X=0
30 FOR I=0 TO N
40 LET X=X+I
50 NEXT I
60 PRINT "X= ";X
70 END
```

Note: On my daughter's Sinclair, using 10,000 for N, the program runs in about a minute and a half and gives an exact answer. Be prepared to wait for a while if N = 1,000,000

The slow speed at which this simple program runs on a personal computer would discourage one from using an 8-bit processor based computer for computing.

Fraud abounds in the computer business. Many people claim that the IBM personal computer is a 16-bit computer. Wrong! It is an 8-bit computer and runs as slowly as any other toy computer. It uses an Intel 8088 processor chip and I quote directly from the Intel component data catalogue page 7-25, "(8088) 8-bit HMOS microprocessor." I also quote from the Intel iAPX 86,88 users manual, page 2-1, "the 8088 is designed with an 8-bit external data path to memory and I/0 —." Any salesman who tries to tell you the IBM PC is a 16-bit computer is lying to you. Do you want to do business with someone who lies to you? Actually, as mentioned earlier, the 8088 is a third generation 8-bit processor with the advantage of addressing as much memory as a normal 16-bit processor can. But, it handles data in 8-bit words and has all the other disadvantages of other 8-bit machines. In addition, nobody has taken advantage of the extra memory capability of the 8088 to write a good operating system. The IBM PC runs outside vendor supplied operating systems translated from what had already been done for limited memory machines, and doesn't work much better.

In addition the IBM PC seems to have a bit of trouble trying to compute. There has been a "bug" since day one that affects its math accuracy. Commands such as PRINT 8.9 give readouts of 8.8999. It makes one wish for a $3.99 blister pack calculator from K-Mart for complex calculations such as check book balancing.

We have had problems with the advanced version of Intel's microprocessor circuits too. For the past year we have been attempting to evaluate the Intel System 86/330A, a complete Intel supplied 16-bit microcomputer using the 8086 true 16-bit processor, a $30,000 package. Simple two port matrix algebra problems, that run without any problem on the $200 HP41CV hand held programmable calculator crash and burn on the Intel system. Long discussions with the local Intel distributor, Intel factory rep., and inquires directly to Intel in California have not resolved the problem. Every few months they supply a new and revised version of their operating system, we try the machine again, and it crashes and burns again. We suspect a circuit board or chip level timing problem inherent in their basic product. Since this design is the base for all of their advanced products, we wonder why they are plowing ahead while the basic product will not compute.

Although the IBM PC is not a 16-bit personal computer, there are some true 16-bit computers now coming into the home computer field. In theory, if executed well, a 16-bit machine should run about 10 times faster and have adequate memory for a highly enriched and easy to program operating system. It should do complex math quickly and be an adequate general purpose machine. The TRS-80 Model 16 and the Hewlett Packard Series 200 Model 16 are true 16-bit computers.

Again, looking first at the hardware available, we find there are only a few 16-bit microprocessor chips being made. Intel makes the 8086 chip, which we have evaluated in Intel's own system 86/330A mentioned above, with poor results. Motorola makes the 68000 chip which is used in the Radio Shack and HP machines mentioned above. The problem with the 68000 is that Motorola does not make any support chips for it, such as math processors, I/0 processors, and other related support circuits. Thus each computer maker must re-invent the wheel and supply discrete circuits for this purpose (which may or may not work well and which make the end product more expensive) or attempt to do some of the necessary support as software commands which makes the system run much more slowly than it should. For example, in evaluating the TRS-80 Model 16 with a reasonable UNIX operating system, we found it would execute no faster than a much less expensive 8-bit personal computer, too slow to use to compute with. We understand Commodore is coming out with a new 16-bit machine based on a Zilog Z-8000 processor which is similar to the Intel 8086, and that National is starting to produce a NS16032 microprocessor but it isn't available in finished products yet.

One basic problem with the new generation of 16-bit machines seems to be that everybody wants their obsolete and nearly useless 8-bit programs to run on them, forgetting that their 8-bit software is really grot. There has been marketing pressure to make 16-bit machines with obsolete 8-bit operating systems that run obsolete 8-bit programs. Wonderful, why not just keep your old 8-bit machine? Why buy a new computer whose main claim is that it runs obsolete programs?

Finally, to realize the possible speed of a 16-bit computer, everything in the system must be faster. You need faster memory chips, faster I/0 processors, faster disc drives, etc. Accessory circuits and hardware designed for 8-bit systems just won't do the job. The system will run no faster than its slowest part, and may in fact crash randomly for no easy to find reason if some support parts are not fast enough.

Thus, in our search for a computer we have written off all 16-bit computers too, as they are all based on the third "bottom up" track of development, now based on marketing — 16-bits are wonderful, whether they do anything useful or not.

It appears there is only one computer in the world now available that is based upon the original second track — distilling downward the capability of a mainframe by further large scale integrating of the hardware and development of better operating systems. That machine is the Hewlett Packard 9000. It is a computer that can compute. For your information, there is also only one 8-bit computer that can compute and that is the little HP75C hand held unit. If you are interested in engineering and not game playing, evaluate the HP75C, it is a sleeper.

If you have got to buy a toy computer anyway, please follow this advice:

1. Don't pay for the computer until you actually take delivery. Fraud abounds.
2. Note that the cost of the computer is only a minor part of the system. It may only be $99.00, but the disc drives, printer, interfaces, etc. can make the price of a useful system over $2000.00, and that isn't cheap for a toy.
3. If you want to play games, buy a game player, not a computer. We suggest the Coleco or Intellivision as 90% of the Atari VCS games are trash.
4. Do not buy software for a computer that you cannot list or that you cannot make backup copies of. Software has bugs. Floppies crash. Software vendors come and go. You can easily end up left holding the bag.
5. If you must buy accessories, disc drives, printers, etc, buy the ones supplied by the computer manufacturer. Do not buy off brand accessories as they may not interface or run with your computer, no matter what the salesman claims.
6. Buy only supplies from sources recommended by the computer manufacturer. Never buy off-brand or "sale" floppy discs. They crash and ruin heads.
7. Don't bother with trying to fit more than 64K of memory into an 8-bit machine. As previously discussed, it isn't directly addressable. Get a disc drive instead.
8. Don't buy any program until you have had the opportunity to first read all the documentation and you can understand it, until you have had the opportunity to sit down with the program and documentation loaded into a duplicate of your machine, complete with all required accessories for that program, and have run thru every sample problem and application for that program and have assured yourself that under all circumstances it works for you and does what you want.
9. Don't believe salesmen promises. Believe only what you can get the machine and the program to actually do. Don't believe the existing bugs "will be fixed in the next revision." Salesmen don't have crystal balls. If its got a bug that gets in your way, don't buy it.
10. Understand the limitations of the machine and use it within those limitations. Don't pay for expensive accessories that do not make the computer better. A Sinclair is a nice $40.00 basic learning device. It is not a word processor, no matter how much extra expensive trash you buy for it. If you want a $5000 word processor, buy one, don't expect to make a toy computer into one.
11. If text editing is important to you, make sure you only consider computers with an 80 column display and both upper and lower case as a standard feature.
12. Good luck in your adventures in computerland. You are going to need it.

Frank Van Alstine

(with lots of help from Aado Perandi)

[1990 Note: We did purchase the HP-9000 computer (actually we got it on a 5 year lease) and did develop our own circuit analysis software that lead to the development of the Transcendence preamp and amplifier circuits and to our patent on them. However, time marches on and Hewlett-Packard stood still. Because the machine was expensive and in limited production, no rational general business software became available and as toy computers grew up over the next five years, by the time our lease was up it was obvious that inexpensive consumer level computers had started lots of companies writing lots of much better software for them. The consumer market and the sophistication of products to fill that market snowballed much faster and further than I had expected when I wrote this in 1983. Thus when the lease was up, the HP-9000 was retired and replaced with a Mac II (now upgraded to a IIfx). Storage and memory cost an order of magnitude or two less than they did 7 years ago and finally many off the shelf programs are available that do useful work – such as Word and PageMaker – which I am using to edit this after scanning it into the Mac with OmniPage. The IIfx is doing our circuit analysis work these days, but still with our own in-house written software using Mathematicia as a framework. Off the shelf programs still are not accurate enough to provide exact answers. We wrote our own bookkeeping and record keeping programs too, under HyperCard. The Mac is used for circuit card layout design, catalog design, writing Audio Basics, circuit analysis, label making, and sometimes, for fun. I wouldn't be without it now. It will be interesting to see what the next 7 years brings. FVA]