AUDIO BASICS

The Complete 1982 Back Issue Set.

VOLUME ONE NUMBER ONE JANUARY, 1982

The purpose of *Audio Basics* is different. Do not expect arm waving subjective reviews of the latest and greatest audio equipment. That would be a conflict of interest for *Audio Basics* in any event, as Jensens Stereo Shop designs its own audio electronics. Write for our catalogue separately if you desire. Our goal is to attempt to teach, to give you some insights into the real world problems with audio components, to separate out the mystic from the objective, to give you enough data and ideas to help you make better decisions with your audio equipment.

We will attempt to explain, in this and in subsequent volumes, those issues not properly covered in either the commercial or underground press. We will normally cover one basic topic per issue, as this is a low cost newsletter, not an attempt at producing a grand and glorious magazine. The topic for this issue is speaker systems, and specifically, how to make most of them perform better (not different, but really better) at a very low cost.

How to Take Advantage of a Little Known Law of Nature and Make Your Loudspeakers Behave Better, Almost for Free.

"For every action, there is an equal and opposite reaction," from the *Principia*, 1686-1687, Sir Isaac Newton, author, (and still true).

What does this fact mean to you, and to your loudspeaker? First we must consider what your loudspeaker really is. It is, in fact, a linear (hopefully) motor mounted on some kind of supporting frame. The AC signal from the amplifier interacts with a magnetic (or electrostatic) field, to cause the movement of a diaphragm, which in turn moves air. Refer back to Mr. Newton. The action of the linear motor (woofer, tweeter, electrostatic diaphragm, etc.) of moving air, causes an equal and opposite reaction of moving the loudspeaker system parts (cabinet, mounting panel(s), woofer framework, etc.) themselves. This isn't nice.

We shall now recite one of Van Alstine's laws of audio, relating to loudspeakers: "In a loudspeaker system, those things that are supposed to vibrate should vibrate exactly as they are supposed to, nothing else should vibrate at all." Unfortunately, Mr. Newton screws us up as mentioned above by insisting that the action of moving the diaphragm (the part that is supposed to vibrate) causes the reaction of vibrating everything else too. Guess what, vibrating loudspeaker frameworks, mounting panels, metal horn throats, magnet assemblies, etc., do not sound very nice. B&W Loudspeakers of Worthing, England, recently came to this same conclusion and studied the vibrations radiating from the cabinet of their 801 loudspeaker system mid-range enclosure, using a laser interferometer and computer to accurately record and document the experiment. They found that the worst case cabinet vibrations were 50 dB lower in output than the acoustic output of the speaker diaphragm. The 801 is a very solidly built speaker and this measurement shows that its spurious resonance was 100,000 times lower than the desired output from the midrange cone. Not a problem you think. Wrong! B&W designed a stiffer midrange enclosure yet (made of concrete) and lowered the unwanted vibrations by another 10 dB, to 1,000,000 times lower than cone output. The results, subjectively, were unbelievable. The improved speaker sounds like a higher quality amplifier is connected to it. It is clearer, less "colored," is "quicker," and smoother. Needless to say, the concrete midrange cabinet is now in production retrofittable to the original B&W 801, and the 801F is in use worldwide by major classical recording studios as the most accurate monitors available.

"But what does this have to do with my loudspeakers?" you ask. "I don't have a laser interferometer and can't change my speaker enclosures to concrete." No, you can't (although we have one client that did!). However, there is a lot you can do to your speakers yourself, following the simple instructions below. We shall take advantage of another basic law of nature, "Meatballs don't bounce." — i.e., there are substances available of high internal friction, that when vibrated, tend to damp out and absorb externally induced vibrations, rather than re-radiate them in unwanted fashion. A good test for a vibration damping material is to make a ball of it, and drop it on a hard surface. If it bounces, don't use it. We have found one substance that works particularly well. It is inert, cheap, non-toxic, non-corrosive, easy to find, easy to apply, dissolves in Ronsonol Lighter Fluid if you make a mess, sticks to metal and wood, and like a meatball, doesn't bounce. The substance is Plast-i-clay, available for about $1.00 a pound, at many toy, drug, department, and dime stores. Depending on your speakers, you will need 5 to 10 pounds of the stuff. Read on before buying it to insure that your speakers have a chance.

STEP ONE: Make sure your loudspeakers are built so we can help them.

A. Is your grill (grill cloth) removable so you can see the actual drivers (woofer, tweeter, etc.)?

B. Can you physically remove the speakers from the cabinet without damaging the speakers or the cabinet? We must have access to the back sides of the loudspeakers themselves.

C. Our suggestions may void your speaker system warranty. If you screw up, you may void your speaker system. Do you want to take the risk?

D. Can you remove the insulation material from your cabinet (if fiberglass, use rubber gloves) to gain access to the bare cabinet walls themselves? You will later have to reinstall the insulation.

E. Since removal of the speakers will require the detachment of the wires to them, you may need to know how to solder, and how to mark wires so you get them back in the same place as they originally were. Can you handle this?

If all answers are yes, proceed as instructed below. If you are not sure, don't do it, the work isn't worth damaging your speakers.

What we propose to do is a simplified version of what B&W did in building a concrete midrange cabinet for the 801F. Although we do not have a laser interferometer, we know your speaker cabinet does in fact resonate. We also know the resonances from it are wrong. We furthermore know that it is possible to damp out some of these resonances with the previously described Plast-i-clay. The resonances will then be less, the system will be "less wrong," and inevitably, "sound better."

You can conduct your own subjective experiments as you proceed, perhaps doing only one speaker system (or only one driver) first, and comparing it with the other speaker. We expect you will be surprised with the results. We must make a few assumptions. We assume that the designer of your speaker system, in general, knew what he was doing and that the selection of drivers themselves, crossover points, and the basic tuning of the cabinet for raw bass response is competent. The following damping will not change the frequency response, Q, crossover

Published each month by Frank Van Alstine of Audio by Van Alstine, Inc., 2202 River Hills Drive, Burnsville, Minnesota 55337 USA. Telephone 612 890-3517. Subscription price for 12 monthly issues is $16.00 per year plus $8.00 per year for foreign air mail. Annual back issue sets available from 1982 at $15.00 per year.

points or slopes, or cone non-linearities of your system. We will be removing small signal errors (spurious resonances) not large signal errors. There is a remote possibility that the suggested damping may make your system sound worse. This can happen if the original system construction is so rotten that spurious output is a substantial portion of the overall sonic output of the system. In that case, the designer may have balanced the crossover and drivers unknowingly to compensate for the abnormally large resonances. Obviously, if many of the resonances are then removed, the balance of the speaker system may then be quite strange. Of course the speaker was a lost cause in any event. We hope you start with reasonable quality speakers, not department store specials.

Now is the time to purchase several pounds of Plast-i-clay. Unwrap a quarter pound stick and knead it in your hands for a while to warm it up (a return to kindergarten.) We will be applying a layer of this material to almost everything inside your speaker system, one quarter to one half inch thick. If you own a micro-wave oven, a pound of Plast-i-clay in the oven on high for one minute will soften it to a nice working consistency.

Now, on to the project, which will use up a Saturday afternoon.

Decide if you want to damp the entire inside of the cabinet, or just the driver frameworks (which takes a lot less work and helps a bunch in any event.)

1. If you want to do the whole speaker system, remove the woofer to gain access to the inside of the cabinet. Note that a few low cost systems have removable cabinet back panels. If so, do it the easy way and remove the panel. Obviously, we want to go about this the easy way.
2. Remove the internal insulation if possible, and save for future reuse exactly as it was originally.
3. Apply a 1/4 to 1/2 inch thick layer of Plast-i-clay to the entire inside surface of your speaker cabinet.
4. Reinstall insulation as it was originally, but over the Plast-i-clay treated surfaces.

Driver damping. This technique applies to all conventional speakers, including mid-range and tweeters. There are two cautions:

l. Do not interfere with the mechanical motion of the diaphragm or its suspension. Do not get any damping material on the speaker cone, surround, or spider.
2. Do not block any air vents. Some speakers have small holes in the back of the magnet. Others have holes or slots in the side of the framework, where the magnet joins the basket (framework). These vents must remain open or the frequency response of the speaker will be altered.

Coat the entire metal framework of the speaker with a layer of Plast-i-clay about 1/4 inch thick. Coat the entire magnet assembly too, watching out for air vents as mentioned above.

1. Coat the entire outside surface of peizo tweeters (such as used in Dahlquist DQ-10s).
2. If your speaker uses horns (such as Klipschorn midrange and tweeter units) then coat the entire outside of the metal horn, but do not coat the inside, as this would interfere with the acoustics of the air flow in the horn. Coat the outside of the driver unit at the rear of the horn too.
3. If your speakers are mounted with brackets, such as horn mounting brackets in K-horns, coat the brackets too.
4. If you own Dahlquists, coat not only the speaker frameworks, but both sides of all the mounting panels, their support brackets, and the outside top surface of the woofer cabinet too. Regarding Dahlquists specifically, they will look ugly, but sound better, if you use them with the large metal grill screen off, as the grill itself has resonances we can't do much about.
5. Carefully examine the surface of the speaker that is mounted to the cabinet. Usually it is worthwhile replacing any original gasket with a thin layer of Plast-i-clay before remounting the driver, and mounting the driver firmly, but not "air wrench" tight on the cabinet surface. This will tend to slightly de-couple the driver from the cabinet, preventing remaining resonances in the driver framework from spreading to the cabinet.
6. Further improvements can be made to dome tweeters and mid-range units by very carefully applying a l/16 inch thick layer of damping material on the front surface of the mounting panel, tapering it at the inside to provide smooth dispersion of the acoustic dome itself, and being very careful to keep the material off of the driven dome or suspension. The vibrations in the tweeter are conducted into its metal mounting plate, where they radiate in a most unmusical way.
7. Does your system contain a passive radiator? If so do not overlook it, its metal framework radiates spurious resonances too. Damp its framework the same as if it was an active driver.
8. If your system has a tuned port, coat the outside surface of the port tube. (Not the inside surface, we don't want to disturb air flow characteristics.)

This completes the mechanical work on your speaker system. In a future issue, we will discuss possible electrical improvements (crossover and wiring ideas.) When you play your speakers now, we expect you will hear that something is missing, namely a lot of garbage that was wrong, and with any luck at all, the improvements to your system will be more than worth the $12.00 you spent on *Audio Basics*.

There is one further mechanical improvement that was shown to me by a client of mine in Barbados, West Indies (where some of the world's best audio systems exist). My client has wrapped a massive concrete slab, about the size of the top of his speaker system, in leatherette and places the slab on top of his speaker. Again, Newton strikes, and the added mass stabilizes the entire system, with cleaner and deeper bass, and better definition and imaging throughout the entire range of the system. Do it, you will like the results.

That's it folks, we hope you like this little newsletter and that it is of value to you. Next month we will be talking about tonearms and turntables, what they do, what they are supposed to do, and what you can do about it.

I would like to hear from those of you who actually take our advice and damp the mechanical resonances in your loudspeaker systems. We will print a few of your comments in a future issue of *Audio Basics*.

Thanks for your attention,

Frank Van Alstine

VOLUME ONE NUMBER TWO FEBRUARY, 1982

Greetings, dear readers, there are about 100 of you out there now. I thought you might like to know about the feedback I have received regarding the first issue of *Audio Basics*, and speaker damping.

Unfortunately, the issue has been received with an incredible silence. I have had a total of two (2) responses. One was a letter asking if the damping would work on a LS3/5A. Of course. The second telling me it was a very bad idea to put damping material on speaker cones as that would add mass to the moving system and screw it up. Evidently writer number two can't read, as I specifically told you to NOT get any of the damping material on cones, suspensions, or any moving parts of the system. Evidently not a single one of you has actually tried my suggestions in issue 1. That is a shame. Thus, there is not much point in showing you how to damp tonearms and turntables at this time, until I actually get some feedback from you indicating that a few brave folks will actually try my suggestions. So, I am changing the subject of *Audio Basics* for this issue to:

How to Read an "Underground" Audio Magazine.

Once upon a time a group of interested researchers decided to "subjectively" test a grasshopper.

They placed the grasshopper on the floor and all shouted, “Jump” at it. The grasshopper jumped all the way across the room. The researchers then pulled all three legs off of the left side of the grasshopper and placed it on the floor again. They then all shouted, “Jump” again. The poor grasshopper only jumped a few inches, and at that mostly sideways. Then the researchers removed the remaining three legs from the grasshopper and tried again. They all shouted, “Jump” several times. The grasshopper didn’t move at all! The results of this test were then published in an underground hi-fi magazine, “Grasshoppers without legs cannot hear!”

Unfortunately, subjective reports with the same validity of logic as grasshoppers without legs cannot hear are published all the time. I mean, of course, cause and effect errors. I have no doubts regarding the ability of golden ear reviewers to hear. I have grave doubts regarding their ability to relate what they hear to what is actually going on objectively in the audio system. And, of course it is impossible for a golden ear to subjectively evaluate audio electronics and then assign perceived defects to any specific engineering shortcomings in the unit under test. In fact it is not uncommon for a golden ear to assign a defect to a unit under test, when, in fact, the perceived defect is elsewhere in the system, and only revealed by the unit under test.

Let’s assume, for example, that great audio reviewer has a pet “reference” system he is happy with. In that system he has a phono cartridge he thinks is just wonderful. He also has a “reference” preamplifier he thinks is just wonderful. Let us also assume, for the sake of this argument, that the cartridge actually isn’t quite perfect, but has a bit of transient overshoot at very high frequency, as almost all cartridges do. In other words the cartridge will add a bit of “zing” to highs. We will also assume that great reviewer’s pet preamplifier is vacuum tube, with all the objective non-linearities this implies — severe current limiting under transient conditions driving a load. Thus the preamp, in fact, slightly limits, especially on high frequency transients. The end result being that the preamp erases the “zing” from the cartridge. Now consider the plight of the poor audio manufacturer who submits a preamp to this great reviewer for evaluation. Said manufacturer by some inspired work of genius having actually created an absolutely perfect preamplifier (don’t worry, no one has, yet). At this point you know what kind of review this preamplifier will get, “This preamplifier isn’t musically acceptable, the high end is harsh. Only vacuum tube preamps can be musically natural.” Have you seen this kind of logic in print recently? Grasshoppers without legs cannot hear.

To put it another way, consider Van Alstine’s Law of Compensating Non-linearities: “Two wrongs in an audio system do not make a right, they make two wrongs. However, the combination of two wrongs may be less obnoxious than either of the separate wrongs individually.” This brings up an interesting question. Is your goal in audio to find that combination of wrongs that leaves you with an audio system that is not obnoxious, or is your goal to learn what the actual objective non-linearities and wrongs are, and to eventually eliminate all of them, at least to the level of human perception ability?

“I saw a cat. It was gray. Therefore, all cats are gray.” “I tried a new amplifier in my audio system. It sounded wonderful. I liked it better than any other amplifier I have tried in my audio system. Therefore this amplifier is wonderful.” Do you see any connection between these two logic statements (both flawed)? Do you see any connection between them and the way audio component reviews read in underground magazines? Let us consider what, in fact, an underground reviewer can objectively tell you in a subjective review, in the absence of objective measurements.

When a subjective reviewer evaluates a component in his “reference” system, he most certainly has the right to decide whether or not he likes the results. He most certainly has the right to write a long essay of purple prose describing his subjective feelings of the way the unit under test behaves sonically in its interaction with the rest of his system. He finally has the right to tell you whether or not he “likes” the results, and how these likes or dislikes compare with other units he has subjectively evaluated. He does not have the right to then assume, and then tell you, that the unit under evaluation was either “good,” “bad,” or any degree between.

Why not? Because, great reviewer has no evidence at all as to whether the unit is good, bad, or indifferent. He only has evidence as to whether or not he likes it. That brings up another point. What does great golden-ear reviewer like?

Once upon a time there was a gentleman who decided to subjectively evaluate food, and meat in particular. He traveled around the world ordering various meals, and finding all sadly lacking. Then, one day he happened upon a White Castle, and ordered one of their (then 12¢). “White Castles.” The White Castle burger is certainly unique, and the gentlemen liked it better than any other form of meat he had ever evaluated. He then wrote a long article for the *Michelin Guide*, suggesting a Three Star rating for White Castle, explaining that the White Castle burger was the “best” meat dish in existence. To his dismay, his article was rejected. Understand of course that the reviewer did have the right to like White Castles better than anything else. He also had the right to claim this in writing. Michelin also had the right to reject his article. Our problem, in audio, is that great underground reviewer usually publishes his own magazine. There isn’t anyone to reject his article. What does great golden ear reviewer like? Will you like it too?

Beware the "Different is Better" Syndrome.

Recently word has been going around in the underground press that the bass and mid-range performance of the Hafler DH-200 amplifier can be made “just wonderful” by direct coupling the input to its circuits (shorting out its input coupling capacitor). The golden ears claim wonderful blossoming bass, great mid-range concert hall ambience, and all sorts of mystic qualities for a DC coupled DH-200 amplifier. Recently one of our clients brought in a DH-200 so modified for us to evaluate. Inasmuch as we have this nasty objective habit of first evaluating the unit on the test bench in accordance with a set of objective standards we have developed that seem to separate the linear from the non-linear (more about these tests later), we dropped the modified DH-200 into the test set-up before risking our speakers (and ears) with it.

Sure enough, DC coupling the input of the Hafler made a profound change in its internal linearity. It made the amplifier much, much worse than stock. The amplitude of 60 Hz ripple correction signal inside the feedback loop was an order of magnitude greater than stock. DC balance could no longer be adjusted to be stable. A low frequency step pulse signal drove the amp completely DC offset with many cycles of decaying oscillations occurring after the input signal was removed. In other words, the amp was turned into a low frequency underdamped tuned oscillator, with all internal devices running in a much less linear transconductance mode. The sonic result was obvious to rational listeners. The bass didn’t “bloom,” it boomed. The mid-range didn’t have “ambiance” it had resonance. The amplifier didn’t “reproduce” music, it created something, certainly interesting, but sadly, not music. Does golden ear reviewer have the right to like this creation? Certainly. Does golden ear have the right to say it is good, or better than stock? No. Because it objectively is not, it is worse. And, I submit, the wonderful sonic effects of this underdamped tuned resonator are absent in the reality of live music.

Needless to say, after our client was able to observe what the actual objective effects of DC coupling his Hafler were, he re-installed the input coupling capacitors, as he no longer “liked” the “bloom” and “ambiance” that modification had created. This does make us wonder what the golden ear likes, and would he still like it if he knew how it was really behaving.

The "Wonderful Today, Trash Tomorrow" Syndrome.

Not long ago, one underground magazine rated the Grado Signature IV as “God’s gift to audiophiles.” This sold lots of cartridges, in spite of its (then) high price. The next issue of that magazine reviewed the cartridge again with an article entitled “A story of True Grit,” totally neurosing all the poor suckers who had just run out and blown several hundred dollars on the cartridge. Is the Grado a competent cartridge? Yes, especially if terminated into 10,000 ohm input impedance. Did the Grado change from issue to issue of said magazine? No. What did change? The magazine's perception of the Grado changed. Which issue of the magazine had the "correct" subjective review? Your guess is as good as mine. Did the magazine ever try running the cartridge into a proper termination? Probably not. Does this make a difference in its subjective sound quality? Of course. Does the magazine care? Who knows? Can they provide "absolute" reviews? Do our perceptions change? Mine do. Even though I have a very competent audio system at home, sometimes I don't like it as much as at other times. I seem to enjoy it best late at night. Is this because my attitude is more relaxed then, or is it because the AC power line may be "cleaner" then? I don't know. I do know that I cannot expect 100% consistent subjective judgements from myself, and I doubt if I can from others. Certainly it does not surprise me when an evaluation of something in an underground magazine changes.

We suggest you keep the following thought in mind when you read that your "latest and greatest" audio component has just been relegated to the junk pile by a review of this month's "latest and greatest." What you have has the qualities that it has, which probably have not changed. Did you like it when you bought it? If so, it probably isn't too bad. A review of something new and wonderful doesn't change the characteristics of what you now own at all. Only your perceptions of what you have will change. And make sure that what is new and great is really new and great, not just new and different. Remember that good designs last. Bozak Concert Grands, 30 year old Klipschorns, AR3A's (if driven properly) remain competent, reasonably linear speakers. Many more steps sideways are made than steps ahead in the search for the Holy Grail of audio.

Magic Capacitors and Foo-foo Dust.

Claims have been made in print recently that "changing all the capacitors in the signal path in your audio electronics will make it sound much more wonderful." Ho ha! Lets do it! Lets pull all the original capacitors and replace them with ones of the same marked value but of much more expensive quality than the original. Lets now listen. Oh wow, isn't that wonderful! Sorry, no, the results are only oh wow, isn't that different. It seems there is a slight flaw in the test procedure. We hope that you would agree that it would be nice, when conducting a scientific test of the sonic quality of capacitors, to have only one variable in the experiment, namely the value of the capacitor. It seems that the definitive tests of capacitor quality recently published have two variables, one unknown to the testers, namely the value of the capacitor. The actual value (capacitance) of a capacitor is NOT what is printed on it. It is what it actually is, which can be measured with a precision meter. Capacitors are manufactured and sold with a certain set of specifications. One of the specs. is the tolerance of the capacitor (how close its actual value comes to the value printed on it). All other things being equal, the tighter the tolerance specified, the higher the cost of the capacitors, and the cost escalates rapidly as very tight tolerance capacitors (1%, for example) are produced by measuring and culling them from a much larger production run. This does not determine quality, only a guaranteed tolerance. It would be much like telling an orange grower that you wish to buy only those oranges from his tree that measure exactly 2.75 inches in diameter. You pay for the cost to measure all the oranges to find the ones your size, and you get very expensive, but not better, oranges.

In audio use, 10% tolerance capacitors are adequate in most circuits, 80% tolerance capacitors are found in many power supplies, and only in specified frequency shaping circuits, such as RIAA equalization, are really tight tolerance capacitors necessary. A reasonable assumption is that most of the capacitors in the signal path in the unit up for magic capacitor testing will be no better than 10% tolerance. So what happens then when you pull a 10% tolerance 8200 pF capacitor, for example, and replace it with another "higher quality" 10% tolerance 8200 pF capacitor? The original 8200 pF capacitor absolutely will not actually be 8200 pF. In fact it is almost guaranteed it will be somewhere in value from 7380 pF to 7790 pF or from 8610 to 9020 pF. Why? Because all the 5% tolerance capacitors or better have already been culled out and sold at higher prices to someone else. Thus when you swap capacitors, you randomly change the value of the capacitor, typically by 10% and easily make a 20% change in value from one channel to the other.

Changing capacitor values in a circuit changes the frequency response and phase response in many cases. One characteristic that we seem to hear, and that helps determine the "where" something is, is the difference in arrival time of a sound source to our two ears, — the phase difference. Thus, it would be nice to have absolutely the same frequency and phase response on both halves (channels) of our audio system. Minor changes in phase response between the two channels are audible. When we replace all the capacitors in our equipment with so called higher quality ones, we make many phase and frequency response changes from channel to channel. The results of course are audible. But, have we made it "better," sorry, no, we have only made it different (unless you are changing capacitors in one of Jensens Stereo Shop units where the original capacitors have been matched channel to channel for exact phase gain balance — then you will absolutely make things worse by destroying this tight match).

Never mind, the Emperor's New Clothes Syndrome strikes again. A change was made (to magic capacitors that we "know" must sound better), and we can hear a change in the equipment. Therefore it must be better. Therefore magic capacitors are wonderful. This is really nice, especially for those who sell magic capacitors.

We suggest the following procedure be used in conducting magic capacitor experiments. As each capacitor is removed from the circuit, measure it with a precision meter (a Data Precision 938 such as we use is good to .05%, which is probably close enough). Now select a new magic capacitor of exactly the same measured value as the one removed. I hate to tell amateur experimenters the bad news, but you are going to need a stock of a few hundred of each of each printed value of magic capacitor to assure a good match which may make your controlled experiment a bit expensive. To save you the cost, I will tell you the results in advance. Assuming that none of the original capacitors were actually defective or used in an improper application (a polarized capacitor in an unbiased centerline coupler application, for example) there is no sonic difference at all between the original capacitor and the new magic capacitor. This result (yes, we have done the controlled tests) is no surprise for those who understand the electronics of audio design. The passive parts are not where the problems are. There probably are problems in your system, large objective measurable problems. They are, however, large, non-linear distortions in the electronics (topography related) and even larger distortions in the mechanical components of the system (cartridge-arm-turntable and speakers). Worrying about the quality of your capacitors, assuming that the originals are not defective, is much like worrying about whether your car's hubcaps should be silver or gold plated, while overlooking the four flat tires. You are directing your worries to the wrong problem.

While discussing capacitors, let's clear up a couple more points. First of all, "high slew rate" coupling capacitors are nonsense. A capacitor properly selected and properly biased in a coupling application DOES NOT SLEW. It does not charge or discharge at all in operation, except in biasing up at the turn on of the unit. It behaves much like a two man push-pull saw the lumberjacks use to cut down our redwood forests. Although both men can do work using it, the saw does not change state at all, unless the men get out of time and bend or buckle the saw. Likewise, a coupling capacitor does not change state unless the circuit design is incompetent. If your electronics need "high slew rate capacitors" you have problems.

One "expert" has claimed that polarized capacitors are bad. This is like saying that a Mercedes-Benz is bad. And it is, if your application is to attempt to carry twenty tons of crushed rock in it. A polarized capacitor is bad too, if improperly used, for example, in a centerline coupling application with no biasing voltage across the capacitor. Then the polarized capacitor will essentially short out on each negative swing and have asymmetrical output. This is bad and will "sound" bad. In an unbiased centerline application, a non-polar capacitor is required, one that can handle both positive and negative potential at each terminal. If properly biased on, a polarized capacitor works just fine. The problem is that many audio designers don't seem to know this. The original Dynaco St-400, for example, has many polarized capacitors in non biased centerline applications. No wonder there has been much arm waving about its "sonic quality" in the underground press in the past.

We also suggest you be very careful in the use of polystyrene capacitors. They have three problems that can get an amateur in real trouble. First of all, they are very sensitive to heat when the leads are soldered. They are much easier to damage (they melt) than most transistors or diodes. Second, their capacitance value changes substantially with temperature. A polystyrene is not the part to use in a tube circuit or near a hot running transistor or resistor. And finally, they change value with rough handling. A hard squeeze on the body can take them out of tolerance. We once saw a Southwest Tiger amplifier in here for service. It had been "modified" with high quality polystyrene capacitors. Every single capacitor was shorted. The amp was a mess. The modification changed it, it ruined it.

Stop the Press, a Letter Came Today Regarding Speaker Damping!

Dear Frank Van Alstine

Re: Jan. "Audio Basics"

I applied 20 pounds (10 each) Plast-i-clay to my Tannoy Belvedere enclosures and am only 1/2 way through. These are big (5 cu. ft. vol.) tuned port (tunnel) cabinets housing Tannoy 15" drivers. Ten lbs. each was only enough for front and back panels, and the tunnels. Needless to say, I have not yet begun to apply clay to the speaker frames and magnet covers, these are pretty massive and "dead" as is, but I suspect I will get further improvement when I do. Even with what I've done so far the improvement is dramatic. It takes a long time with the big surfaces, but well worth it.

Subjectively:

1) Less awareness of the bottom end "being there" as either too muddy or too thin; it just blends in better and with sharp reduction of bass decay time.
2) Less sense of strain and overload throughout.
3) "Room effects" seem less, imaging better. I decided not to trade my speakers after hearing them with Grado - Connoisseur arm and MOS-FET 150, SUPER-FET Pat-4. The clay damping is a further refinement and I've only just begun.

Regards,

Roy V. Varner, M.D.

OK people, refer back to issue one of *Audio Basics* again, buy that Plast-i-clay and get going. Why did you subscribe to this little newsletter?

Obviously I have touched upon many subjects in this issue that may raise questions and touch a nerve here and there. I need your feedback to cover things of general interest in more detail. I also need to know how you perceive what I write, for it isn't what I write that is important, but how you perceive it. The communication isn't any good unless you receive the message.

There are lots of things hinted at in this issue to be covered in the future. Such as: What kinds of objective distortions are really occurring in my system? How can they be objectively measured? What do they "sound like?" We will cover arm and turntable damping in the future. What about "Digital" sound? What is really happening with digital? Why don't some people like it? (Rest assured, the forthcoming Compact Digital Disc is just fine, and its problems are inconsequential in relation to the gross problems of the analog record and its recording and playback process). In our reference system, in which we have reasonable documentation as to the objective behavior of each component, it is possible that one of the worst case remaining problems is the analog tape recorder used to make most of my records. Certainly the sonic "signature" of compression of transients so common to all my analog tape source records doesn't seem to be there on any of the direct discs or digital tape source records. We have also had the good fortune to operate our Transcendence preamplifier and Transcendence 400 amplifier on direct digital master tapes, at Soundstream in Salt Lake City last fall. It doesn't take much of an ear at all to realize that the digital masters are of higher quality than any record. Needless to say, we will not be among those writing to CBS (as one underground has urged), to lament about the horrible quality of digital and how it destroys music, because it doesn't. In every age Luddites abound.

Until next month,

Frank Van Alstine

VOLUME ONE NUMBER THREE MARCH, 1982

I thought you might like to know there are now over 140 of you that have paid $12.00 for this newsletter. At this rate of subscription increases, we will surpass the circulation of *Reader's Digest* in only 14,000 years. I too will then have to worry about whether or not to accept advertising.

We got lots of feedback this month. It was along two general lines. 1. Those who did try speaker system damping with Plast-i-clay as described in issue #1 (lots of you) and liked the results. 2. Those who claim to like "magic capacitors" in spite of my comments. All I can say is that you do have the right to "like" magic capacitors. I might note that an editor of *Audiogram* called me to let me know that he "likes" our new Transcendence preamplifier better than any other he has heard. We don't use any magic capacitors in it. I wonder how then it is possible for him to like the preamp? Anyway, on to the topic for this issue.

Record Playback Systems, What They Do, What They Don't Do, What You Can Do About It, and Finally, Since the Compact Digital Audio Disk is Coming Soon, Is It Worthwhile to Worry About the Mechanical Record Playback System Much Anyway?

WHAT A TONEARM IS (Van Alstine's definition): A tone arm with a cartridge mounted, while playing a record, is a multi-frequency vibration generator on the end of a stick. This is not what you would like a tone arm to be. However, Van Alstine's corollary to Murphy's Law: "Mother Nature Don't Give a Damn" applies. Another way to put it is that everything, without exception, behaves exactly in accordance with the physical laws of the universe. This may not be the way you would like something to behave, but that's tough. We would like a tone arm to be an absolutely inert, absolutely rigid, support for the cartridge, with infinite mass, infinite length, and yet be able to track warped records perfectly. We would like it to be supported so it has no play, no slop, and no friction. We would like all mechanical energy generated by the mechanical following of the record groove by the stylus assembly to be converted into electrical energy, with no energy lost, no energy added. We are not going to get our "likes." However, since most tonearms do not even come close to our goals, and since the closer we get to what we would "like" the better the playback system seems to behave, there are a few obvious things to do to your tone arm – cartridge that seems to help.

To consider how bleak the situation actually is, think about the amount of play (slop) there is in the bearings of your tone arm. Consider the amount of slop in relation to the actual size of the information on the record groove. Consider that you have, in fact, an information retrieval system, in which the amount of uncertainty in the system is greater than the amplitude of the information you wish to retrieve. In other words, its like trying to read a page of Braille with a sledgehammer. Its kind of amazing that the system works at all. The following is a crude description of how the system actually works – not a description of how you would like it to work. The stylus assembly attempts to follow the imperfectly formed record groove. Some energy (signal) is lost in deflecting the vinyl, some is lost in wearing the vinyl, some unwanted energy is gained as the vinyl bounces back. More unwanted signal is added from mechanical noises from the turntable. If you have a dust bug or record brush mounted on your cartridge, more unwanted energy is inputted into the system from it. Part of the energy deflects the diamond tip in relation to the cantilever and is dissipated before getting to the coils. Part is reflected back to the vinyl again (bad). The cantilever, which should be absolutely rigid, isn't. Thus it behaves more like a garden hose than a high speed lever, some energy is transferred along it in ripples and again reflected back to the diamond tip and the record. What does get to the other end of the cantilever is partially dissipated as heat in the suspension. Some finally is translated into electrical energy by the interaction of a coil in a magnetic field and is sent merrily on its way to your preamp. Some energy, however, goes into moving your cartridge coils (even if you don't have a moving coil cartridge). Some induces tuned resonances in the hollow tuned resonant cavities in your cartridge. Some energy moves your entire stylus assembly in relation to the body of the cartridge, and some energy is dumped into your tone arm which rattles (or wobbles) around in its bearing assembly, dumping part of the energy back into the cartridge again. Mechanical vibrations from your turntable also dump energy into your tone arm, providing additional bearing rattle or slop. At each parts interface (cartridge to headshell, headshell to arm tube, arm tube to bearing assembly, etc.) part of the energy is reflected back, part continuing on to rattle things downstream. If your arm is complex, built of many parts, with lots of "dingleberries" at the bearing end, the unwanted spurious resonance modes are nearly infinite.

If your arm is a unipivot, some energy is lost rotating the arm, screwing up imaging and deep bass. If your arm is a conventional gimbaled type, it actually sets on the bottom bearing and rattles back and forth in the gimbal. Got a knife edge bearing? They go scrape, scrape, and rattle, rattle. Of course the amplitude of all these aberrations is tiny. However, the size of the information you wish to retrieve is tiny too. You do have a big problem. And you thought all you needed was magic headshell wires and the right Vertical Tracking Angle!

D0 YOU HAVE ANY PLAST-I-CLAY LEFT (from damping your speakers)? Although we cannot effect a complete cure, the judicious use of a little Plast-i-clay can help the situation quite a bit, and improve your tone arm (appropriately named) nearly as much as it helped your speakers.

A few minor skills are required, such as being able to re-balance your tone arm to the proper tracking force for your cartridge. In addition, it would probably be nice if you started by reading the instructions that came with your arm again (if it was a separate component) so you know what adjustments can be made to it for arm height, anti-skate, etc. In addition, re-check to see that it was properly installed on the mounting board in the first place (most we see to install cartridges in here are not). Do not believe the tracking force and anti-skate gauges built into your arm. Buy a separate stylus gauge. The little plastic AR balance is just fine.

Lets Start with the Cartridge.

1. De-garbage your cartridge. Some phono cartridges come with lots of extra dingleberries attached directly to them, such as clip-on brushes and hinged stylus guards. You have enough trash rattling around without having to cope with these too. If you can remove built in stylus guards and dust brushes without damaging the cartridge, do so.

2. Almost all magnetic cartridges have removable stylus assemblies. Many are built along the lines of a Shure with a long metal tube that slides into a hole in the body (Shure, Stanton, Empire, Pickering, Nagatron, Grace, etc.). Most of the rest have "clip-on" stylus assemblies (ADC, Sonus, Signet, Audio-Technica, etc.). Grado cartridges are unique, and this section doesn't apply to them as Grado stylus assemblies are already firmly located as supplied. Obviously, this section does not apply to moving coil cartridges without user replaceable stylus assemblies. The common characteristics of all cartridges similar to those mentioned above is that there is substantial "slop" in the fitting between the removable stylus assembly and the body. Since the cantilever is in the removable section, and the magnetic receiving section is in the body, any uncertainty between the two sections causes a large error (most obvious at low frequencies). You can actually feel this error (slop) by gently wiggling the stylus assembly in relation to the body (on Shure type cartridges). This slop can be substantially reduced with a very small amount of Plast-i-clay.

3. All you have to do is to pack a small amount of Plast-i-clay into the "cracks" – the junction between the body and the stylus assembly. Obviously, avoid getting any material into the suspension. A toothpick is helpful, and cotton swabs are useful for wiping off excess material. The Plast-i-clay will both "firm up" the interface, and help damp out spurious resonances at the same time.

4. Its time to play your cartridge again. Since you have probably changed its mass, (either by removing dingleberries and or adding a bit of Plast-i-clay) you will obviously have to re-balance your arm. Note that no cartridge performs best at the manufacturer's minimum recommended tracking force. That is advertising hype. If, for example, the spec. sheet claims the cartridge tracks at "between .7 and 1.5 grams," a good starting place would be 1.25 grams. Remember, for best performance, the generator end of the cantilever must be centered between the pole pieces when the cartridge is on the record. If the tracking force is too low, in addition to some mistracking "shatter," the cantilever will not deflect enough, and the generator will not be centered. Likewise, too great a tracking force will deflect the cantilever too much, again uncentering the generator – pole piece alignment, causing asymmetrical output.

Anyway, we expect that your cartridge will now play cleaner and less "boomy" deep bass, and exhibit improved imaging and cleaner highs too. It might make you wonder why the cartridge manufacturers don't do it right in the first place, its so easy.

Electrical Interface of Cartridge with Preamp (or Receiver).

As long as we are discussing the cartridge first, we might as well go over the importance of a proper electrical interface between the cartridge and the phono input of your preamp or receiver.

The operational characteristics of your phono cartridge can be separated into two distinct classes. The first are the mechanical characteristics. This class includes the stylus and suspension and its mechanical motion and resonances, and the mechanical resonances inherent in the body structure; the mechanical frequency response of your cartridge as limited by the compliance of its suspension, the mass, length, and stiffness of the cantilever, the polish of the diamond and its alignment, all mechanical dips, peaks and resonances. It is the mechanical output of your cartridge as further modified by tone arm and turntable resonances that is transformed into the electrical signal.

The electrical characteristics of the cartridge can then be considered separately making a few reasonable assumptions:

1. An ideal cartridge should have a flat mechanical response.

2. An ideal cartridge should have a flat electrical response.

3. It is not likely that mechanical non-linearities can be compensated for by designed in electrical non-linearities.

4. It is likely that mechanical and electrical non-linearities are additive, producing sonic problems worse than either apart from each other.

5. It is probably a good idea to make the electrical frequency response as flat as possible, no matter what the mechanical response of the cartridge may be (one problem is probably not as bad as two problems).

The electrical characteristics of a magnetic cartridge are easy to consider. We have only a simple electronic circuit consisting of an inductor (the cartridge coil) in series with a resistor (the DC resistance of the coil) in parallel with a capacitor (the distributed capacitance of the tone arm and interconnect cable) in parallel with a resistor (the input impedance of your phono section). Stray capacitance in the cartridge body and the DC resistance of your interconnect cables can be considered to be negligible in comparison to the value of the circuit components mentioned above. This circuit configuration is easy for any competent electrical engineer or technician to recognize. It is a 12 dB/octave tuned resonant circuit, sometimes called a “Tank” circuit. The circuit is flat at low frequencies, and at some high frequency, if properly executed, will start to roll off at 12 dB/octave. If improperly executed, at high frequencies the response will rise before it rolls off, causing a high frequency peak in the response. Whether the circuit is flat or not is completely dependent on the value of the components in the circuit.

Obviously, for any given brand of phono cartridge, the values of coil inductance and coil DC resistance are fixed for that particular cartridge. Remember too, that each different brand of phono cartridge has a different value of coil inductance and DC resistance. There is no industry standard for these values. There is, in fact, over an order of magnitude difference in coil inductance and resistance between various cartridge brands. The capacitance of each tone arm brand is different too, although over a lesser range (about 2 to 1). The capacitance of interconnect cables also vary from about 100 pF to 300 pF depending on type and length. (Note: Price and magic qualities of interconnect cables have no correlation with the real qualities of the cable).

In general, the greater the coil inductance the lower the frequency at which the cartridge starts to roll off. The greater the interconnect cable capacitance, the lower the frequency at which the cartridge starts to roll off. The smaller the value of the internal coil resistance, the greater the amplitude of any high frequency peak. And finally the higher the terminating resistance is (phono input impedance) the greater the amplitude of any high frequency peak will be. Obviously then, adding “matching capacitance” across your phono input adds to the interconnect cable capacitance, and causes the cartridge to roll off at a lower frequency. If the values of coil inductance, cable capacitance, and termination resistance are too high, the high frequency roll off will occur well within the audio range. A typical “problem case” magnetic cartridge could then be described as one with high coil inductance (over 500 mH) used with high capacitance interconnect cables (over 300 pF) and terminated into a high load resistance (40 KΩ or more). This cartridge would have a substantial electrical peak in the 11 to 14 K Hz range and then “die” not covering the top end of the “audio” bandwidth at all. Unfortunately, many “popular” magnetic cartridges do indeed have these “problem case” characteristics. Check the specifications for your cartridge.

Lets backtrack to the previously mentioned “Tank” circuit again. Remember that we informed you that the higher the terminating resistance, the larger any high frequency peak will be. We do not want a high frequency peak in the electrical response of our cartridge. Preamps don’t like high frequency trash (the phono stage overloads). Power amps don’t like high frequency trash (they slew limit). Tweeters don’t like high frequency trash. Even if this high frequency peak is well above the “audio” range, it causes problems that screw up the sound in the audio range. For example, if any stage of your system slew limits at any frequency, all signals entering your system during the time any stage is slewing are erased – or good-bye definition. The problem is curable.

We cannot change the internal values of coil inductance and DC resistance in your cartridge. The only way to do this is to purchase a different brand of cartridge. Interconnect cable capacitance can only be changed by acquiring lower capacitance cable (a good idea). Obviously, any “matching capacitance” is bad as it lowers the frequency of the tuned Tank circuit. There is, however, one circuit element that is easy to change, the input impedance of your phono preamp. This is effected by installing the proper value resistor in parallel with your phono inputs.

For any given cartridge and turntable, in which the values of inductance, DC resistance, and cable capacitance are “fixed,” there is one, and only one, value of load resistance that causes the electrical characteristics of that particular Tank circuit to be optimally flat. Simple electronic circuits are real neat. It is possible to calculate the proper circuit values. The calculated value will always work exactly right. If we calculate the proper value load resistance, the circuit will then always measure flat on the test bench, and surprise, it will always sound best in the system. Therefore, to insure that the electrical response of your phono cartridge is flat, with no high frequency peaks, all that is required is to calculate the proper value load resistor and install a resistor in parallel with your phono inputs so the combined parallel resistance of the resistor and the standard 47,000 ohm input resistance equals the desired load for that cartridge.

Two concepts need to be pounded in here: A. The “industry standard” 47,000 ohm load is wrong! Since each cartridge brand is different, the required load for each cartridge is different, and a 47,000 ohm load is random, and wrong for all magnetic cartridges. B. The problem could be cured if either the cartridge manufacturer was smart enough to design his cartridges to work properly into a 47,000 ohm load, or if the user is smart enough to change the 47,000 ohm load to that proper for that particular cartridge.

For those of you that are interested, the following is the mathematical formula for solving the problem of what is the proper load resistor value for your particular phono cartridge. It will give you the pole point (-6 dB frequency) of your Tank circuit when the circuit is critically damped (Q = .5). It will also give you the value of the load resistance required to achieve a critically damped Q with your cartridge. You will need to know the following factors: l. The inductance of your-cartridge coils. 2. The DC resistance of your cartridge coils. 3. The total capacitance of your tone arm wiring and interconnect cables. Note that you must add to #3 the internal input capacitance of your preamp, including any “matching capacitance,” if it is significant in relation to #3. If you solve this equation for a cartridge with high coil inductance, you will notice an interesting set of problems: A. If the cartridge is critically damped, it will have no high frequency output in the audio range. B. If the cartridge is used with a 47,000 ohm load it will have a large peak in the audio range. Neither choice is proper. There is no other choice except to replace the cartridge with one having much lower coil inductance. Sorry about that.

Let R = Series resistance of cartridge coil

L = Inductance of cartridge coil

C = Cable capacitance

R_L = Load resistance required to achieve critical damping. (Q = 0.5)

$$R_L = \frac{-RLC + 2L\sqrt{LC}}{4LC - (RC)^2}$$

RL/47K = Load resistance required in parallel with 47KΩ to achieve R_L. Note that R_L must be less than 47KΩ.

$$R_L//47K = \frac{47x10^3 R_L}{47x10^3 - R_L}$$

f(-6dB) = -6dB pole frequency

$$f(-6dB) = \frac{\frac{L}{R_L} + RC}{4\pi LC}$$

If you do not care to do the mathematics yourself (it can be handled on any simple calculator) we will do it for you for $5.00 per phono cartridge. We have become tired of doing it for free with “thank you’s” few and far between.

However, as a general guideline a load resistance that is in the ball park for many cartridges is as follows:

10,000 ohms: All Grado cartridges

24,000 ohms: All Sonus, Nagatron, low inductance Empire, and others with coil inductance in the 200 - 300 millihenry range.

30,000 ohms: Grace, Ortofon, and other cartridges with medium coil inductance values.

47,000 ohms: You might as well use the standard input impedance with high coil inductance cartridges inasmuch as they cannot be critically damped and flat in the audio range.

47,000 ohms: All active head amps

500+ K ohms: All moving coil transformers. This is only possible if the inherent input impedance of your phono-section is high and not feedback dependent so that the nominal 47 K ohm load resistors can be removed. This is possible to achieve with our SUPER-FET and TRANSCENDENCE preamplifiers, but not with many others.

??? Moving coil cartridges: A. In general, the interface problem is between the cartridge and the head amp or transformer. B. In general, the coil inductance and DC resistance is so low that the tuned resonance occurs above 100 K Hz and may be damped by the internal resistance of the interconnect cables or transformer primary. C. Manufacturers of moving coil cartridges, transformers, and head-amps usually supply so little objective data that inadequate information is available to solve the equation.

To install the proper load value in your preamp it is necessary only to solder the appropriate value resistors (one per channel) between the phono hot and phono ground lugs of your phono input jacks inside the chassis of your unit. 1/8 watt 5% carbon film resistors (available at Radio Shack) work just fine.

One final thought for this month. Since the load resistance of any given cartridge makes a profound change in its sonic quality, all "reviews" of phono cartridges that do not first take care to have each cartridge operating into a critically damped Q are totally useless. The "golden ears" will be listening more to trash caused by underdamped operation of the cartridge than to the actual inherent sonic quality of the cartridge. Subjective opinions of what is the "best" cartridge are worthless in an uncontrolled environment. The "grasshoppers without legs cannot hear" syndrome strikes again.

Sorry folks, I have run out of space for this month. Tonearm damping will be continued next month.

Frank Van Alstine

VOLUME ONE NUMBER FOUR APRIL, 1982

Hello again dear readers, we have made a new invention this month, one that can profoundly affect the quality of your record playback system. Please pay attention to this month's *Audio Basics* and do go ahead and try the "do-it-yourself" phono cartridge modification herein described. We suspect it will shock and surprise you.

Because we consider this invention to be very important in any playback system, and because it is not an invention that we feel can be patented (or hidden – for reasons that will become obvious later in this issue), we believe the best thing we can do with this invention is to publish it, so all will know it is our idea, and perhaps convince a few people that we do know something about audio design.

Thus we are sending a copy of this issue only to all phono cartridge manufactures in the U.S.A., all commercial and "underground" magazines, and to a variety of "high end" hi-fi stores. It will be very interesting indeed if we get any reaction or feedback at all from these people. Inasmuch as the "NIH" syndrome (Not Invented Here), is very strong in this industry, I will be very surprised if I get a response, either positive or negative, from over 1% of those in the "professional" end of this business.

To further the possibility of making the herein described phono cartridge modification available to as many interested audiophiles as possible, I hereby give permission for this issue only of *Audio Basics* to be reprinted by any audio trade journal, underground magazine, audio club newsletter, commercial audio publication, and cartridge manufacturer's newsletter with the specific condition that the reprint contain our name and address and subscription information for *Audio Basics* ($16.00 for 12 monthly issues to Jensens Stereo Shop, 2202 River Hills Drive, Burnsville, Minn. 55337).

The "LONGHORN" modification is, essentially, a bit of "mass management" designed to add a high polar moment of inertia to your cartridge without increasing overall mass any more than necessary. My cartridge builder, Robert Nielsen, came up with the idea after pondering the Koetsu cartridge and a tonearm stabilizer for a while. In our opinion, the main difference between the Koetsu and other moving coil cartridges is its high mass. It is dense! Sonically, it seems to have better dynamic range and transient performance than most other cartridges. This is not surprising, for all other things being equal, the greater the mass of the cartridge, the harder it will be for the cantilever to move the cartridge body instead of generating the electrical audio signal.

Unfortunately, the very high mass of the Koetsu is not without drawbacks. It is too heavy to balance in some tonearms, and its performance on warped records leaves much to be desired. In fact we are a little suspect of how long it will stay in good playing condition as its suspension is "worked" so hard over warps. Its price is a little outrageous too.

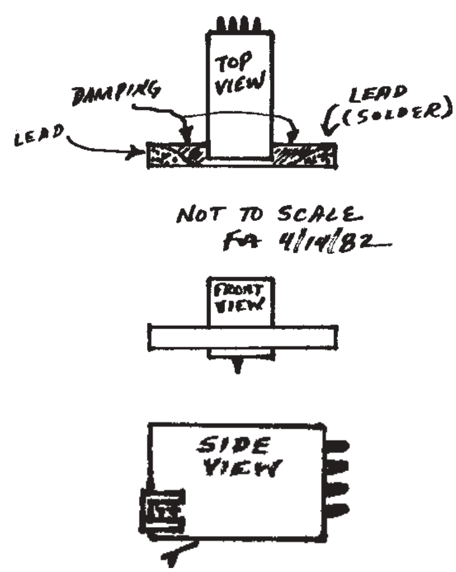
We have recently evaluated an interesting device that clamps onto the tonearm. It is essentially a bar extending out about 1.5 inches on each side of the arm with an "outrigger" weight on each end. The object of the device is to stabilize the arm by adding a high polar moment of inertia. This device (Final Audio Research Cartridge-EQ KKC-48) (what a complex name for a simple device) works just fine, improving the image, bass response, transient response, and dynamic range of the cartridge – arm combination. It too has drawbacks. It has lots of mass, the overhang is in the way of putting records on and off the turntable (sooner or later you will scratch a record with it), and the polar moment is not added in the best possible place – below the center of gravity of the system and very close to the cantilever.

Bob Nielsen's idea then is obvious. If adding a polar stabilizer to the arm is helpful, and if the inertial resistance of the high mass Koetsu body is helpful, then why not do it right, and universally, by attaching an inertial stabilizer bar directly to the cartridge itself? Add the inertial resistance in the right place, below the center of gravity of the system and very close to the cantilever, where it does the most good with the minimum amount of mass increase.

The rest of this issue of *Audio Basics* tells you how to do it yourself, almost for free, and improve your phono cartridge about five hundred dollars worth.

CAUTION: There are three drawbacks to the "Longhorn" modification.

1. The stabilizer bar may interfere with a record weight or clamp. If you have to make a choice, we suggest that the Longhorn stabilizer is much more effective than a record weight or clamp.
2. If your cartridge is already on the heavy side, the addition of the stabilizer bar may increase the mass beyond the limits of your tonearm's balance range. The combination of a massive cartridge and very compliant suspension may have adverse effects on suspension life and warp performance. Note however that we would rather have a cartridge that plays 95% of our records perfectly than one that plays 100% of our records, but plays them all poorly.
3. It is necessary to remove all "dingleberries" – brushes, hinged stylus guards, and other



trash from the front face of your cartridge. If you cannot do this without damaging your cartridge then do not proceed. Obviously, the modification will work better if you have previously bonded the stylus assembly to the body as described in the March, 1982 issue of *Audio Basics*.

The Longhorn Inertial Stabilizer Universal Cartridge Modification

MATERIALS REQUIRED:

1.5 inch long section of K&S #181 1/8" brass U channel (available at most hobby shops).

1 tube DURO DEPEND adhesive (to bond brass U channel to cartridge). *[1990 Note: Use 5 minute epoxy glue now. Depend did not hold up well enough.]*

Solder pencil, rosin core solder, (the ability to use same).

A small amount of Plast-i-clay or Permoplast (the same thing) or DOW CORNING SILICONE RUBBER SEALANT (to damp the U-channel).

Small metal file (to file notch in U-channel to fit your cartridge body).

Steady hands, (woopses bung cartridges) steady eyes, (to get it on straight) and enough ambition to try it.

A small vice is handy to hold things while filing or gluing.

Overall Procedure and Purpose:

The modification will add a lead (solder) weighted brass U-Channel across the face of your phono cartridge, mounted as low on the face as possible, which, if mounted properly, will be parallel with the record surface and below the center of gravity of the cartridge arm system. It will provide a high polar moment of inertia to mechanical input into the cartridge (vibrations from the record grove) and will cause more signal to be converted into electrical output and less wasted in mechanical movement of the cartridge and arm. The subjective result is distinctly better separation and imaging, better bass, better transient attack, better dynamic range, and less mistracking.

Directions:

1. Go to local hobby shop and buy a section of K&S #181 brass 1/8 inch U-Channel (or similar). Pick up DEPEND, metal file, and whatever else you need at the same time.
2. Saw off a 1.5 inch length of the U-Channel. File a notch in the U-Channel the width of the face of your cartridge body (or stylus assembly – whichever is appropriate). The notch should be centered, and about 3/4 of the depth of the U-Channel. The "open" end of the U will point back at your tone arm bearing end. File and trial fit until the bar is a snug fit onto the face of your cartridge. About 1/2 inch of bar will protrude out on each side of the cartridge.
3. After you have a good trial fit, fill the outer one-half of each end of the bar with solder flowed in with a hot solder pencil. (NOT with the U-Channel attached to the cartridge – not to melt things! !)
4. After things cool off, fill the rest of the U-Channel (except at the notch) with Plast-i-clay or Silicone sealant to damp the bar.
5. For a "trial performance" tack the finished Longhorn bar to the face of the cartridge with a bit of Plast- i-clay. Remount and re-balance the cartridge in your arm and give it a listen. Note that lots of things have gone away (trash). Note that lots of new things are there (details and dynamics).
6. If you like the results, permanently mount the Longhorn bar with Duro DEPEND, trying to get it on straight, and fill in the remaining "cracks" with damping material.
7. Use the left over U-Channel to make stabilizing bars for all your friend's cartridges. Be the hero of your local audio club. (But make sure you tell them it was our idea).

This is enough basic audio advice for one issue. We would be delighted to print your reaction to the "Longhorn" in a future issue of *Audio Basics*. We will continue with tone arm damping in the next issue (honest).

Frank Van Alstine

VOLUME ONE NUMBER FIVE MAY, 1982

Sorry that we are a week late this month. I spent most of last week in Nashville, Tennessee, showing off our Transcendence 400 amplifier to recording studios there. Its an interesting place, studio after studio with more electronics, controls, knobs, and switches than any starship will ever require.

The people in the recording industry there are sharp, and are trying to put out the best sound that they can. Sometimes they get carried away (in much the same way a "mid-fi" audio hobbyist does) and use layer after layer of equalizers, compressors, expanders, noise reducers, and every other signal processor known to man (and some unknown) to attempt to get the "right" sound. It was kind of fun to watch their jaws drop when I set up one of our little MOS-FET amplifiers driving only two little full range monitor speakers directly off their tape machines, and it outplayed their complete bi-amped and tri-amped studio systems. I mentioned at one studio that their reference 400 watt amplifier was not playing bass. They told me that there wasn't any deep bass on that master tape. Then we hooked up the Transcendence 400 and there was plenty of solid deep bass!

So I asked them the same question I ask you: What is the use of expensive electronic crossovers, bi-amping or tri-amping, and "sub-woofers" when your power amplifier doesn't play bass at all (If you have a conventional solid state or vacuum tube amplifier you have the same problem)? Anyway, I am happy to report that people in Nashville can hear just fine, that they do care about their product, and that they do want to produce the most musical record they can. The Transcendence 400 is wanted back for an extended audition. Another interesting eye opener occurred at the main stage of the Grand Old Opry. We compared the Transcendence 400 with their best 400 watt per channel stuff and not only did the Transcendence 400 drive their equipment into the ground, it provided coverage over a much greater area off axis than their equipment. How would you like to go to a concert using amplified music and have the whole thing just sound like super clean music, not like a P.A. system? I think we have got the Nashville folks thinking.

Word of what we are trying to do with *Audio Basics* seems to be getting around. Subscribers number over 200 now. Keep it up and I will have to acquire a computer to keep track of all of you, but not an 8 bit system, they are all toys and inadequate for business purposes. If we can talk Intel into a decent deal on their new full 16 bit System 86/330 with a 35MB Winchester, 1 megabyte of main memory, and iRMX 86 operating system we will have a computer that will do real work for us.

Feedback from *Audio Basics* #4 (the LONGHORN cartridge modification).

I guess the feedback could best be expressed by one reader who informed me that he thought that the Longhorn modification was the single most important improvement he had ever made in his audio system at any price. I have received about a dozen letters from readers expressing this same opinion. If you have not done the Longhorn to your phono cartridge yet, you don't know what your music system sounds like at all! I find that if we go back to a cartridge without the Longhorn, the system sounds so terrible I cannot stand to listen to it. It makes me wonder how it is possible to evaluate the "sound" of a preamp or amplifier at all when the source is bung.

I stopped by a high end audio salon in St. Louis recently with a Longhorn modified Grado cartridge (the one we sell for $125). I made an interesting request – I asked them what was the worst turntable and arm they had in the place – one so bad they were embarrassed to sell it. They referred me to one NAD model that they claimed was so harsh they couldn't sell it. I mounted the Longhorn Grado in it. You should have been there! It then outplayed everything in the shop, it was so amazing that the two owners started laughing, they couldn't believe it. We were getting more musical highs, better dynamic range, better balance, cleaner bass, better imaging, and just plain much more musical sound from the little NAD turntable and the Longhorn Grado into a cheap NAD receiver and B&W DM10's than they could get out of their very most expensive systems. The difference wasn't subtle – it was obvious. Needless to say they bought my sample Longhorn Grado on the spot and ordered more. I left the owners feeling more optimistic about their ability to deliver quality sound then they had felt in two years.

I also visited Gary Vart of *Audiogram* with the Longhorn Grado. He really didn't want me to set it up, as his "good" turntable and Fulton arm were down with bung arm bearings. The only other tonearm he had in the house was a SME-IV which both he and I "knew" sounded very harsh and lumpy. I mounted the Longhorn Grado in the SME anyway. To the astonishment of both of us, the SME "vanished" and the Longhorn performed as it does in every tonearm I have tried it in – with utterly superior musicality, dynamics, and imaging. This leads me to an interesting conclusion:

You do not need a new turntable or tonearm – all you need is the Longhorn. It really doesn't matter what your turntable or arm is (as long as it isn't a direct drive "shaker table"). I recently set up a Longhorn Grado in a cheap (but good) Technics SL B-202 for our field rep in Long Island. He then compared it to their Oracle turntable with Fidelity Research arm and Koetsu Black cartridge. The Technics/Longhorn Grado flat wiped it out. They were stunned, our setup which cost less than 10% of their reference (we sell the package for $225 delivered) turned the Oracle/Koetsu into shambles. Save your money for the soon coming Digital Audio Disc player, all you need to get the best out of your collection of analog discs is a Longhorn and the turntable and arm you now own.

In issue #4 of *Audio Basics*, I mentioned that I was sending a copy of the Longhorn modification to all cartridge manufacturers, audio magazines, and a large selection of "high end" audio stores. You might be interested in the response. From commercial audio magazines - none - the information was dropped into another black hole. From "underground" magazines – two responses – *Audiogram* tried it and to say they like the results would be an understatement. *Sensible Sound* responded that the idea sounded interesting and that they might try it sometime. As expected, no response from the other underground magazines. I suspect they are too busy listening to their bung $1000 moving coils to try anything inexpensive that works. Tube God said "thanks for the newsletter" (that's all?).

We heard from four cartridge manufacturers. AKG called me and the importer asked a bunch more questions and said they would try it – and sent us a bunch of AKG literature. Nagatron wrote to say that the idea sounded interesting (but they have not tried it as far as I know). Micro-Acoustics wrote too, and their national sales manager subscribed to *Audio Basics*. Since the Micro sales manager used to be with Sonus, he wondered if the idea worked better with a Sonus or with a Micro-Acoustics. I suggested that he try it and find out. Finally, this week a "surprise" package showed up from England. A Mayware MC-2V moving coil direct from Mayware for us to affix the Longhorn and return – they even paid for our service! I had not even sent them the newsletter – a friend of the company had sent them a copy and they were interested enough to go ahead and send us a Mayware cartridge to rebuild. The NIH syndrome isn't everywhere. I will let you know next month how the Longhorn works on the Mayware MC-2V. (*Audiogram* says it helps all of the moving coil cartridges they have tried it on).

The most disappointing response was from the over 100 audio high end shops we sent the Longhorn information to. Only one response, from Underground Sound of Memphis, Tennessee who said the Longhorn cured mistracking problems with their Grado cartridges. It is kind of funny, all these so called "high end" audio salons, whose very livelihood absolutely depends upon them selling you high quality audio equipment, and only one in a hundred that had enough interest to try the best thing they could possibly do for your audio system. Again, I suspect they are too busy trying to sell the expensive trash to want to try something cheap that works – they wouldn't make enough money on it, so it couldn't be good.

Note that our efforts to "give away" the Longhorn idea is definitely not altruistic. It is in our best interest for you to have the best possible music source. The more you like your system, the more people who come to realize that music can be reproduced with reality, that high fidelity means just that and not just another gimmick or "thing" to own as a status symbol, the more people there will be that are interested in our electronics. Now, for the most part, audio promises, but does not deliver what it promises, and is turning people off. Leisure dollars are going into video, games, and gimmicks. Music reproduction can approach reality. We are trying to help your system do that and turn audio back on again.

At last – as promised – turntable and tonearm damping!

First of all, it is necessary to understand how your turntable and tonearm should work, if they were perfect.

Ignoring cartridge alignment and location for the time being, we can consider that your turntable and tonearm setup has three basic parts; the platter and its bearing, the tonearm and its bearings, and the drive motor for the platter and its bearings.

Obviously, it would be nice to keep all mechanical vibrations that are generated in the motor isolated from the platter (Consider that this is rather difficult to do in a direct drive turntable, in which the motor is the platter – or – do you really think its a wonderful idea to set your record and stylus assembly directly on a running electric motor?).

Anyway, it would also be nice if the motor had no vibrations, and if the platter bearings were perfect (no slop, no play, no friction, no wobble, no noise). It would also be a good idea if all vibrations generated in the motor and platter could be kept out of the tonearm structure. However, it is also important that the tonearm structure be exactly and precisely located in relation to the platter (remember that the record is supported by the platter, the cartridge by the arm structure and thus any relative motion between the platter and arm structure is translated into a signal error by the cartridge – which is real stupid and doesn't know where the signal is coming from). Thus an isolated cushion between the arm and mounting board is not a good idea – it may reduce noise coming in from the platter and motor, but it gives an imprecise location to the arm and generates more error signal than it removes.

Consider also that the whole shebang: arm, platter, and motor, is mounted on some kind of panel (which tends to behave like a sounding board) and the whole works is usually mounted in a box (the base) which is full of tuned resonant cavities. This does not even begin to consider the effects of structural or airborne feedback from the speakers or other outside vibration sources such as you stomping around or even effects of AC power line fields. We haven't even begun to cope with the little problem that the VTA of no two records is the same, that even the best phono cartridge is a lousy device, and that the records themselves are pretty bad. Do you really think that you can drive a mechanical cutter into the original lacquer master with any real degree of linearity? At this point we should just give up and wait for the Digital Audio Disc player, which, by definition, eliminates all of these problems.

Real high fidelity is actually coming folks, just keep calm (That's not to say that someone cannot make a bad digital audio disc player, or use crummy taste in mixing with digital masters – actually the extended dynamic range and noise limits will allow those who really work at it to do a worse job with digital than they can now do with analog – but that's another story).

Do you still have some of the Plast-i-clay (or Permoplast – the same thing) left from your speaker cabinet damping project? Note that Roma Plastilina #1 works just fine too. It is available in 2 pound blocks from Roma Plastilina, 38 East 30th Street, New York, New York 10016. Although we cannot cure all the problems with an analog turntable, we can help out a lot by damping out as many of the internal vibrations as possible.

Start by removing the platter from your turntable, tying the arm to its rest so it cannot bounce around, turning the unit upside down and removing the bottom cover from the turntable base. Caution! With Linn and other tables with oil filled bearings, put a piece of tape over the bearing hole so the oil doesn't all run out.

Now apply a 1/2" layer of Plast-i-clay to all parts of the inside of the base so as not to interfere with the mechanical function of the turntable or with its suspension. Pack the corners of the base heavily. If your turntable has an internal metal frame that supports the platter and arm (such as in the AR or Linn) coat the metal framework too. Apply a thin layer to the inside of the removed bottom panel (watch for clearance with the moving mechanical parts!) and reinstall the bottom cover. You may have to readjust spring tension on turntables such as the Linn to bring the platter back to the correct height and level.

The tonearm structure resonances can be tamed too. Start by packing the Plast-i-clay around the arm base where it connects to the mounting board. It can be heavily "potted" here. A ring of damping material around the tonearm, about 1/8" thick and 1/4" wide, where the arm tube meets the headshell is worthwhile, along with another ring where the arm tube meets the bearing housing, if this can be done without getting in the way of the cueing mechanism.

We also suggest you pack a layer of damping inside the headshell, behind the cartridge. You can "pot" the headshell wires here at the same time if you are careful. The material does not belong between the cartridge and the headshell, the cartridge should be mounted very firmly in the headshell. If you are careful, you can also damp the tonearm bearing structure itself. Being very careful to not interfere with the mechanical motion of the arm and its bearings, it is possible to add a layer of damping to the obviously resonant metal parts such as the outer gimbal, top of unipivot, supports for knife edge, etc. This is a hard one to explain in writing, call us to discuss your tonearm if you desire.

Obviously the slight added mass will require that you re-balance your tonearm now. We suspect you will find that a lot of bad things have gone away when you listen to it again.

The best way to kill resonances in your turntable platter is to purchase the TRI-PAD record mat, made by Eon in Canada and sold by Monster Cable dealers in the U.S.A. This mat is unique. It is a three layer laminate. The bottom layer is black, soft, and inert and seems to do an excellent job of killing mechanical noise from the metal platter. The top layer is a nicely finished layer of cork, which is a stiff and inert support for the record. The entire mat is stiff, dead, and light at the same time. It will not add too much mass for your bearings or suspension to handle, it is thin, and you will not have to change the VTA angle of your cartridge. Yes, we realize that *Absolute Sound* said that it didn't sound good – it made their moving coils sound "hard" in the mid-range. Sorry, their stock moving coil cartridges are "hard" in the mid-range, and all the TRI-PAD did is remove enough other garbage so that their cartridge problems were more audible. People with bad moving coils may like a nice soft, rubbery mat that will change resonances around and hide problems. But the mat must have a Q of .5 or less too (like all other parts of the system) and an underdamped rubbery mat doesn't make things sound better, just different.

Radio Shack has a very effective record clamp available for about $5.00. I understand it is going out of production because it doesn't sell. Average Radio Shack customers don't know what it is, and audiophiles never shop Radio Shack (not expensive enough). Some of my readers tell me it can be improved by gluing a layer of felt to its bottom, but even stock, it seems to work as well as any. It clamps the record firmly to the mat and helps stabilize things a bit more. Buy it soon, it may not be available in the future.

A very effective turntable base can be made by making a quick trip to your local concrete block factory. A standard concrete chimney block is 8" high, 21" wide, and 17" deep and weighs about 50 pounds. A stack of four of them, bonded together with panel adhesive and topped with a slate, concrete, or thick particle board slab makes a turntable support so stable you can kick the assembly while the table is playing, with no effect. Even a single chimney block, setting on a layer of upholstery foam (thick enough so the block almost, but not quite, compresses it) will decouple the turntable from everything except earthquakes.

That's about all I can think of to make a turntable work. Write me if you have some better ideas.

Next month we will give you a report on the June C.E.S. show in Chicago where next year's equipment will all be displayed. You will read about it in other publications sometime next winter if you are lucky. Then we will try and tell you a little about the potential of the DAD (digital audio disc) and how and why it works.

Frank Van Alstine

VOLUME ONE NUMBER SIX JUNE, 1982

This issue is devoted to covering the C.E.S. (Consumer Electronics Show), the enormous dealer only electronics trade show, held each year in Chicago. Inasmuch as there are over 1000 manufacturers displaying the home entertainment equipment they would like to sell you starting next fall, I believe you would like to know what is coming.

The show (held June 6th thru 9th this year) was a complete madhouse as usual. It is not open to the public, but is a gathering from all over the world of electronic manufacturers, representatives, dealers, electronics press writers and editors, and a few celebrities thrown in for good measure. It was claimed there were over 50,000 people in Chicago for this trade exhibition.

The range of "home entertainment" products displayed ran the complete spectrum from the most elite and esoteric $10,000 audio amplifiers to dozens of video tape porno movie hucksters, complete with their X rated "stars" signing autographs for crowding lines of distinguished businessmen. Half of Japan is at the show, and it's kind of an eerie feeling to be crowded into an elevator in the Hilton Hotel surrounded by 20 orientals and not hear a word of English. Kind of makes one wonder who won.

The show is so huge it is divided into three separate buildings, plus a bunch of "private" hotel hospitality suites scattered around downtown Chicago (Mark Levinson and other similar "elitists" do not want us ordinary audio dealers to bother them with evaluating their products, so they "hide" them for their special dealers only).

McCormick Place, the vast exhibition hall on the Chicago lake front is overfilled with the normal "mid-fi" exhibits. The building is so big you could hold a couple of Super Bowls in it at the same time. I think the Goodyear blimp could put on acrobatic exhibitions inside and nobody would notice. Its BIG! The main (top) floor of McCormick Place is filled by huge displays from the major Japanese manufacturers. Can you imagine seeing and hearing everything made by Pioneer, Sony, Kenwood, Technics, etc. all in the same place? Each display occupies about twice the space of a suburban home, and probably costs more to set up. The sound of the hall is what you would imagine if you could turn on every Japanese audio system in the world in the same room at the same time. It is an interesting experience, to say the least.

The front balcony of McCormick Place was a "line up" of every shape, color, and size of satellite receiver dish, all attempting to find Satcom 3 (with little success) thru the microwave "noise" of downtown Chicago. It looked a little like the Martians had landed. By the way, a satellite receiver can work just fine, I have a 10' Channel Master in my back yard and haven't watched network TV in a year, although 90% of what's broadcast via satellite is trash too, its just there is a lot more trash to choose from.

The next floor down at McCormick was a completely different experience. Displayed were every electronic game, video game cartridge, electronic watch, calculator, etc. to go on the market next year. Have you ever been in a southern swamp on a hot humid evening and listened to the noise of all the frogs and bugs beeping? Imagine an electronic swamp with all the beeps, buzzes, creaks and croaks done electronically and you have the sound of this huge show floor. Mixed in were the sounds of all the "lo-fi" manufacturers who unfortunately play their $99.00 compacts, boom-boxes, consoles, and "Pac-Man" radios louder than ever to make up for the gross distortion.

Here and there were the car radio folks, with Porsches, Caddies, etc. right on the show floor playing their “high quality” car audio systems. Sorry folks, there is no such thing as a good car system, there are only very expensive bad car systems and very expensive very bad car systems. They all sound like you were trapped inside a Lloyds radio (shades of Lily Tomlin) with the volume set on full distort. One outfit had even built in two 15" woofers behind and firing into the back seat rest of the car. The bass sounded like you were sticking your head into a 55 gallon oil drum and someone was kicking the other end. And they were all smiling! Why?

Getting the most attention on the second level (after the Atari games and 100 new clones of same) were the porno movie displays, which seems to be big business now. It does seem to be a bit strange that your good old TV and Appliance store can now be your source of dirty movies. Another sickly humorous side note is that evidently the video disc business is actually suffering because porno movies are not available on video discs, only on video cassettes. Thus the big flap was whether Pioneer was, or was not going to start pressing X rated video discs as a service to mankind (last word is they are not). It wouldn’t be so sadly funny if the material wasn’t so bad. Whether or not you object to the material on moral grounds doesn’t change the fact that the “product” is boring, tired, shoddy goods and why anyone would want to pay for this trash is beyond me. If anyone ever made a “good” porno movie they would probably get rich, but probably there isn’t anyone in that business (including the dealers) who knows what good is.

The bottom floor of McCormick Place was filled with the remnants of the CB business, Taiwan AM-FM clock radios (hundreds of them), car alarm systems (all screaming at once), and even a stereo FM wrist watch. Do you really want a “Pac Man” telephone? Someone must. There were myriads of sorry little speakers, all clones of what has been poorly done for the past 20 years (you can buy three way 12" speakers in genuine walnut cabinets from Hong Kong for $20 each — list price $400), such a deal for your good customers. We walked the whole 10 miles of aisles though, for once in a while we find a gem hidden away in the dross. We found Cramolin Red there last year, a marvelous contact cleaner for audio equipment (and your car’s electrical system too!)

As if the above described zoo wouldn’t take more than a week to observe carefully, if one was a masochist, McCormick Place held even more. The once outer lobbies were little refugee camps of prefab tin “listening rooms” set up for the higher class audio exhibits. Essentially, they all sounded the same; boom, boom, boom, or what would you expect trying to demo hi fi in a 10' x 20' metal room. Nevertheless, a few quiet exhibits were interesting and it was a great place to stock up on the latest quality records from Telarc, Sheffield, Nautilus, American Gramaphone, etc. (as long as you didn’t mind carrying a shopping bag of records with you all day). Smart folks bought records last, on the way back to their hotels. One also had to avoid the pretty girls dressed in very little, passing out all the heavy trade magazines until late in the day. David Umeda, my assistant, who is about 5' 2", came across one striking blond trade magazine distributor, who was about 6’5". I nearly lost him for the day with a strained neck. One also must avoid being arm tackled by obnoxious reps attempting to drag you into the most god-awful sound rooms. I got back at them this year, I wore my name badge upside down and spent the day watching reps break their necks trying to read my badge.

It wasn’t all bad. As usual there was some very good and interesting displays at McCormick Place. Many Japanese companies displayed Compact Digital Audio Disc players. In fact, the very best sound I heard at the entire show was by Pioneer (no kidding!). I was lucky enough to walk into Pioneer’s Digital Audio Disc display room when a normal “demo” was not going on and thus was able to hear their system at a rational sound level (almost all users of digital source material were using the equipment to drive things even harder, and further into clipping and gross overload than normal). But used within the limitations of the amplifier power, the absolutely pure source makes music! Sony and Philips both had excellent displays using headphones only, and you cannot believe little headphones sounding that good. Sony evidently cannot stand success though, and in a hotel demo suite, showed their very expensive Esprit “high end” equipment (ever seen a speaker with a square woofer?) driven into absolutely gross distortion and clipping by a digital disc player. When I asked their nerd of a rep to turn it down so I could hear it, he smiled and turned it up further into worse distortion yet. I left, quickly. Yes it is possible to get very bad sound with a digital source, if you are stupid enough, many were.

Several rather pathetic “high technology” displays were at least interesting. Several manufacturers were showing their version of stereo AM broadcasting and receiving equipment. (There is a format fight between several non-compatible systems going on right now). I hope they all lose! Stereo AM, as displayed, sounds like bad stereo FM, who needs it? Stereo sound for TV may have a chance, but those showing systems seemed to be happy with sound quality similar to a K-Mart compact cassette player. (Aado, my electrical engineer, informs me the TV broadcast bandwidth really doesn’t have room for a quality stereo sound that is compatible with existing TV sets in any event). The world does not need more crummy equipment doing more things as bad as ever.

The after the fact noise reduction systems were still trying. I guess they don’t know that digital source material is coming and have not figured out yet that the way to have low noise is to have no noise in the source, not to attempt to take it out later. Sorry, I am not into DBX, Dolby, etc., or any other more or less after the fact noise processors. What I want is a digital disc player, now.

Leaving McCormick Place was a bad mistake, as we left when the show closed for the day and ended up in a line about a mile long for the busses. Would have taken a cab but that line was longer yet. I couldn’t believe it, the line of nice professional people behaved like a mob leaving Calcutta on the heels of a tidal wave. I think people would have killed for a place on a bus. Getting to the head of the line was absolutely dangerous as then you had the real possibility of being pushed off the curb in front of the approaching bus by the thousands behind you. It was not fun.

An overnight collapse after a fill of prime beef (I think its impossible to find a bad restaurant in Chicago) and then back into the fray again, this time to the McCormick Inn, a 20 story high rise motel next to McCormick Place that housed the “purer” audio exhibits in five floors of hotel suites, together with a zillion more speaker companies located in more tin prefab “sound rooms” built on the ballroom floor.

Things of interest: Infinity had a static display of a completely restyled line of loudspeakers, from cheap to the $25,000.00 IRS system. If they sound as nice as they look they should be worth considering. AR was showing off its digital real time “equalizer” with an impressive A-B of a set of large Infinity speakers with the real time equalizer switched in or out as desired. There was one little problem — the Infinity speakers sounded just awful in their room and I have heard them behave very musically elsewhere in good systems. When I operated the A-B switch the sound changed from boomy, lumpy, and bright to nasal, muffled, and dull, with a bit of hollow thrown in. I didn’t have the heart to ask which “way” was with the AR digital equalizer in circuit as both versions were bad, bad, bad. They may be onto something, but different is not better.

After stumbling thru a maze of little loudspeaker demo rooms we found an oasis of music in the desert, the little Celestion SL- 6 speaker sitting there playing so nicely that it couldn’t be missed, even after two days of ear numbing noise. I ordered an evaluation set to see if it stays “real” in my own reference system here. Will let you know next month. Its a computer designed speaker (more or less) with very unique materials and cone construction, a fresh approach at real world problem solving.

Carver was showing his new FM tuner which seems to be able to eliminate multipath noise and make bad stations listenable. I have lots of respect for Carver and his equipment does exactly what he claims it will do. I am a bit suspect that perhaps what he doesn't claim it to do it doesn't do well at all. But, if multipath distortion is a big problem for you, the Carver tuner certainly is the answer.

Micro Acoustics had an interesting static display showing that when loaded into a standard phono input (47,000 ohms and the normal 100 – 200 pF cable capacitance) the electrical transient response of the Micro cartridge was essentially perfect, while a competitive magnetic cartridge rang and oscillated like a bell. The display was accurate, fair, and showed that at least one cartridge manufacturer knows how badly a moving magnet cartridge can behave when improperly terminated electrically. The only thing Micro did not show was that the competitive cartridge would not ring at all if (as we have informed you earlier in *Audio Basics*) it had been terminated correctly for that specific cartridge. Micro's display was fair though, as the magnetic cartridge was terminated into its maker's recommended load, the standard 47,000 ohms. It is not Micro's fault if the other manufacturer is too stupid to know how their cartridge should be used. Are you still running your cartridge into 47,000 ohms?

Looking for a projection TV system? Look no further than the Kloss Novabeam, it is the state of the art system at this time.

Cerwin Vega, who's motto is "loud is beautiful" had access to digital master tapes this year. Sorry, clipping is not beautiful.

Technics had a most strange Digital Audio display. After being trapped into a little sound room, one was forced to suffer thru an audio visual presentation. The visual was on an out of focus and fuzzy rear projection TV system, with audio on a bunch of their crummy little buzzy speakers. The presentation was all incoherent mumblings which could have referred to corn flakes as easily as digital players. Finally a gal switched in a Technics digital tape player using a Mobile Fidelity analog to digital to analog remaster of an original master of Pink Floyd. We were underwhelmed, especially with the Technics DC amplifiers actually playing DC into the woofers — bletcha, bletcha, bletcha. They then finished us off with a Compact Digital Disc of the Planets, again driving their equipment into hard clipping. What were they trying to show?

Philips, in contrast, had a fine professional display of the Compact Digital Audio Disc with excellent sound quality. They have a very high quality new brochure explaining clearly how the system works. Ask your friendly Philips dealer for one, or write to Philips for it at 100 E. 42nd St., New York, NY 10017.

The only major problem with the Compact Digital Audio Disc is that you won't be able to buy one for another year. Although it goes on sale in Japan and Europe this fall, evidently the US record suppliers have dropped the ball and have not aimed towards producing software, hoping if they ignore it, it will go away. Since imported software would be more expensive than desirable Sony, Philips, etc., held a news conference to the effect that the hardware would be delayed until US suppliers got their ass in gear and started tooling to produce the digital records.

We got smart this day and left the show a bit early and got a bus without getting trampled. Now for the most important bit of advice given to you yet in *Audio Basics*: There is an obscure little Italian restaurant in the near north side of Chicago (500 N. Franklin) that consistently cooks the best steaks I have ever had anywhere, ever! At C.E.S. time after 7:00 P.M. the place is a madhouse, packed like the last elevator down from the Towering Inferno. We got in at 5:00 P.M. this year and got a table right away. In addition to the biggest, best flavored, and best prepared charcoal broiled steaks I have ever had, the waiters are real professionals. If in Chicago, don't miss it. Interesting, from the outside the place looks like a dingy little Italian restaurant. Inside, I have never seen anyone order from the Italian menu. We originally found out about the place from a tip from a manufacturer from England. 0h yes, the name of the place is Gene and Georgetti's.

Have we covered the show completely now? Our feet say yes, but there is still another complete building full of displays to go, the Conrad Hilton Hotel downtown, with over 175 rooms full of the more esoteric audio equipment. There were more than a few very unhappy displayers at the Hilton, who, upon first entering their display "suites" said, in effect, "Oh, what a nice entry hall my assigned suite has," only to find out to their dismay that the tiny 8' x 10' "entry way" was their "suite." It was a bit sad watching people try and demo their 4' x 5' $5000 speakers in an 8' x 10' room. C.E.S. better move the high end audio displays to a hotel that has a few more rooms and a lot less closets.

Overall impressions: Acoustat is moving up in the world, their display having grown over the years from tiny hotel rooms, to big hotel suites, to this year, one of the biggest Hilton ballrooms. There is a never ending supply of "hernia maker" vacuum tube power amplifiers coming and going each year. At least the dealer won't need to worry about shoplifters, it takes a fork lift to move most of those slagpiles. I overheard one "ultra esoteric" supplier talking to an "ultra esoteric" west coast underground reviewer, with the reviewer explaining, with a straight face, that she needed to go back and check with another esoteric reviewer for a new supply of his "wonder caps." Same reviewer refused my offer to give her a sample of our $99.00 Longhorn Grado cartridge to play with at her convenience. I sure wish I had ESP too, and could tell, in advance without listening or testing, that a product wasn't worthwhile considering. Are the reviews done the same way too, via ESP? Aren't you lucky to be able to buy this kind of advice – output with no input? Remember, if it isn't too expensive to be able to afford, it cannot be any good. The thought for today, if all the editors of all the magic underground magazines got together and completely rewired all of the electric systems and electronics of a Boeing 747 with magic wires and wonder caps all by themselves, would you fly on that airplane? But its OK in your audio system, right?

A few more general impressions: In spite of really bad rooms, a whole bunch of small companies were getting pretty good sound with a wide variety of equipment. In general, sonic quality seemed to be inversely proportional to cost and size of the equipment.

There seems to be a growing "dead zone" of rationally priced equipment that works well. The mainline Japanese "mid-fi" manufacturers are concentrating on the bottom line equipment (20 watt receivers and cassette decks) while the price of limited production equipment is going beyond the reach of most people. For example, since our supplier of Connoisseur turntables seems to no longer have them available (we are told the company in England has changed hands and production "will resume") , we looked very hard for a quality, simple, and rationally priced turntable. Sorry, there are none. You have the choice of a Japanese mid-fi unit and even though the Technics SL-B202 works very well it will soon be gone too as all the major oriental companies are standardizing next year on units with "P Mount" arms – tonearms ending in a plug fitting directly accepting phono cartridges of a uniform mass, compliance, and length that plug directly into the arm end. This is being done because average mid-fi salesman cannot or will not mount a standard cartridge in a headshell with a good chance of getting it in right side up. Thus the mid-fi manufacturers are eliminating the possibility of the cartridge being set dreadfully wrong, with the trade-off that it will now be impossible to mount it exactly right, and make the tables unusable with any standard mount cartridge that comes along that really is a lot better than what is now being done. One US cartridge manufacturer told me it was a kind of a back door attack to try and take over the US cartridge business too, as has been done with receivers. I was also told speakers will be next. Or you can choose an $800 and up turntable. Even the Rega, which is a rational choice has gone from a $200 to over $400 price. Darn it, it should be possible for someone to produce a good $200 belt drive turntable. Please, won't someone do it, at least Linn could lower the price of the LP-12 to $200 and still make money.

Specific impressions: Acoustat is producing some very formidable speakers, but check your ceiling height! Some of them are 7'9" high! They also are offering a new amp and preamp with clever engineering. An examination of the schematics does make one wonder if they may have solved a few problems at the expense of adding a few new ones. They definitely offer better value than most companies though and thoroughly deserve their success.

PS Audio had their usual intelligent display and nice sound. They used a good Infinity speaker system of known quality so at least one could make a reasonable judgment that their equipment was working well. Many electronics companies came in with unknown bizarre speakers too, so it becomes impossible to tell what is happening. Hafler was showing a new tuner, equalizer, and speaker system, all of which probably have promise. Spica was showing a new unusually shaped speaker that imaged very well indeed. I still like the Dahlquists, although I was underwhelmed by a prototype $8000 plasma tweeter system. Dimension IV showed some very large and ugly (but very nice sounding) speakers. Is there really a market for great big boxes? Esoteric Audio had a display of really expensive tube equipment that did sound musical. (But did those 100+ pound tube amps scattered around the hotel from several manufacturers get their sound rooms hot and swampy! Can you like it if you can't stand to be in the same room with it?) Best Audio Products offered a rationally priced turntable isolation base that seems to work Very well indeed. The Tri-Pad still seems to be a good choice in platter mats, but nobody stocks them, keep trying.

Elite Townsend showed a crushed granite turntable with a center bearing 2" across. In addition, it had a damper being a trough full of silicon located at the front of the tonearm, supported over the record, complete with catchpot to prevent "drippings" from falling on the record. Sounded just fine, but your wife sure won't like it. ACR Industries showed the Apature Trident 3 piece satellite – subwoofer system that seemed as good as any of this type I have yet heard. Inasmuch as the entire system is only $500.00, it certainly is one of the better buys. Try and listen to it. The workmanship on the Spectral preamp is outstanding. I have not heard it under rational conditions yet. Stu Hegeman is with Adcom now, and will be producing his Happi 2 preamp as a nicely done do-it-yourself kit (Hafler needs some competition).

We had a bit of fun as I brought along one of our smaller mos-fet amps in my briefcase and used it to make a more careful evaluation of some of the interesting speakers. We were walking down one hall when I was recognized and was asked (in jest) if I had an amplifier with me as that sound room (Dimension IV) seemed to have good speakers but their "modified" Hafler didn't sound nice at all. I answered that sure I did have an amp in my briefcase, just what they needed, and much to their surprise, popped it out and wiped out the Hafler. At other displays, people seemed to like our mos-fet amp better than the Threshold and Electrocompaniet amps then in use. One speaker manufacturer even gave me an unsolicited order for a couple.

B&W had their usual impressive display (the most professional of the show) with a comprehensive audio visual presentation of the workings of their manufacturing process – a big education packed into a very short time. Their sonic display was devoted completely to their new LM-1 mobile speaker, which is about the size of a brick, denser than same, and plays big and clean. It can go in cars, boats, outside, or in a home system with distinction. Their only problem will be that there are no car electronics good enough for it. From the line up of the world's audio press giving very careful attention to B&W, I suspect you will be hearing more about the B&W LM-1 very soon. B&W ordered one of my amps too.

Allison has completely restyled their speakers and they are now very attractive. I didn't get to hear them but your wife will like their looks. Citation showed a ridiculous $7500.00 amplifier with output connectors probably too big for Fulton Gold wires. It is apparently designed by Otalla in Finland (government subsidized) (buy your socialized amplifier here!). Polk showed a bunch of new speakers with front moldings similar to their "display" version of the RTA-12. Now if they will only make them in production the same as they show them. Several high priced exhibits were using the latest Grado Signature 7 cartridge with little success. It tended to launch itself out of the record grooves intermittently, especially if used in sprung under – damped turntables such as the Oracle. Evidently a Signature 8 is needed. (Our Longhorn stabilizer does cure the problem – I understand Infinity was using one in their hospitality suite – we didn't make it there).

Readers, there is no way I can cover or remember everything, its turning into a big blur after only two weeks. If I missed your favorite brand you are welcome to call me and I will be happy to let you know what I know about any products you are interested in. See you next month.

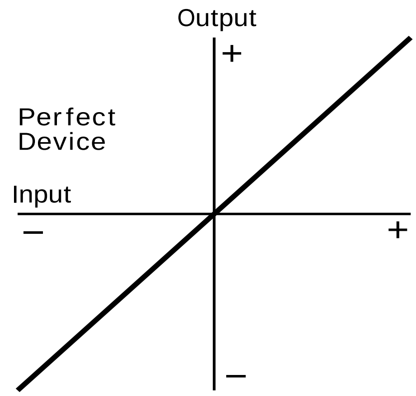
Frank Van Alstine

Volume One Number Seven
July, 1982

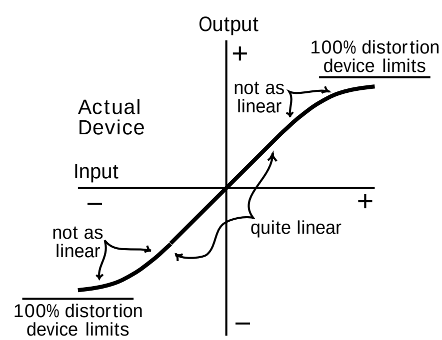
Hello again, I am glad that the feedback I am getting from many of you indicates that you are happy with the information and ideas I have been describing. Some of you have complained that you want more "meat" — technical information on audio equipment. So, this month we will get a bit more technical and go through the operation of a vacuum tube amplifier, what it does, what it cannot do, and what you can do about it. The data will pertain to vacuum tube amplifiers in general, and the Dyna St-70 specifically, as we get so many requests for information on how to modify them.

First though, I want to report on the Celestion SL-6 speaker system I mentioned in Issue 6 of *Audio Basics*. We finally got a sample set this week (Celestion had just appointed a new factory rep for our area and had him bring us a sample set rather than ship directly to us).

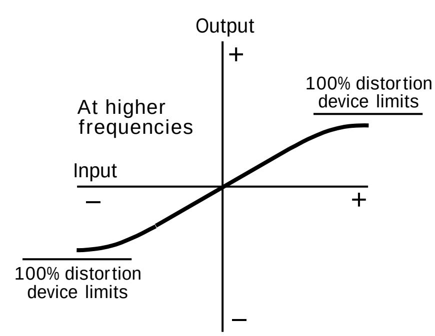
Celestion has, in a nutshell, used similar computer assisted technology as B&W to examine the non-linearities of loudspeakers. Not surprising, both companies can reasonably well document problems that just do not resolve with conventional sweep testing. To simplify, B&W uses their data to model and design drivers of rather conventional construction (but with superior materials and execution) in which the non-linear breakups of ordinary speakers are drastically reduced. Celestion has taken another approach — to 'throw away the book" and design



drivers in which the problems do not exist. Thus the SL-6 woofer is a single piece unit, with no dust-cap (it is an integrated part of the center of the cone), no coil former junction (again, an integrated part of the cone, and no discontinuity at the edge (the surround integrates with the cone edge). The voice coil for the metal dome tweeter is an extension inward of the edge of the dome,



and great care has been made in locating suspension points and lead in wires.



Celestion's ideas work. It is the first speaker we have auditioned here in five years that we could not "tear to shreds" in comparison with a similarly priced B&W model. The SL-6 is seamless, extraordinarily smooth, images beautifully, and has as musical a mid-range and top end as anything we have yet heard, price not being an object. It is however, a little "lumpy" in the lower mid-range and bass, slightly "heavy" sounding under some conditions. We suspect that its cabinet resonances are worse than B&W. Understand that these are not major problems and that at $800/pair it is an outstanding value. If Celestion would have B&W design and build the enclosure for the SL-6 we might get an awesome product. Certainly it is probably a good candidate for a cabinet damping project with Plast-i-clay. I suspect the SL-6 will get mixed reviews in the "underground press." It is so high in definition it will show up harshness in electronics and cartridges that other speakers mask. Many reviewers will end up reviewing their electronics, not the SL-6. One caution, the SL-6 is the only Celestion model built with this new technology. The rest of the line isn't in this class yet. The SL-6 certainly deserves your consideration if you are looking for new speakers. [1990 Note: The SL-6 is now obsolete - think about a B&W CM-1 speaker if you are looking for great and small].

Vacuum Tube Amplifiers

Any audio amplifier can essentially be modeled as a two stage device; a voltage amplifier (which amplifies the amplitude of the signal) followed by a current amplifier (which supplies the drive current to drive the now large amplitude signal into a low impedance load - the speaker system). Although many tube circuits will also have voltage gain in the current amplifier section, that is not important to the discussion that follow. The closed loop gain of the amplifier is determined by a voltage divider which sends a portion of the output signal back to the input of the voltage amplifier out of phase. The amplifier then does not amplify the input signal, it amplifies a signal being the difference between input and feedback signal.

If the voltage amplifier and current amplifier were perfectly linear, the difference between the input and feedback signal would be only a small version of the input signal and the results would be "perfect." In theory, any difference between input and output is supposed to be eliminated by feedback. The feedback signal, which is a duplicate of the output signal, when subtracted from the input signal, should create a "difference signal" which is "pre-distorted" by the amount of distortion in the circuits, but out of phase with the circuit's distortion. The "pre-distortion" in the "difference" signal, when added out of phase to the actual distortion in the circuits, cancels exactly, giving a "perfect" output signal. That is the way it is supposed to work.

By the way, there ain't no such thing as a no feedback amplifier! There are many different feedback schemes. One can choose to use lots of local feedback in each stage and little overall loop feedback, but each device in itself (whether it is transistor or tube) has internal feedback. If one attempts to make an amplifier with "no feedback" except for that inherent in the devices themselves then the design becomes absolutely dependent of the characteristics of each independent device. No two tubes or transistors, even of the same type, are identical. Even if you painstakingly select and bias each individual device one at a time, its characteristics will change with variations in temperature, current, voltage, and age. It will be impossible to make any two channels the same and to keep them the same. The main characteristics of a so called "no feedback" amplifier are: very high cost (each unit is essentially a one off), very hot running as the devices have to be biased on very hard, unstable operation as the unit changes characteristics with age and temperature, no two samples will sound the same as they are device dependent, and lots of "blow-ups." Obviously, repairs will be expensive as a repair is essentially a re-engineering of that unit with new devices and re-biasing of each device. I can live without it. The "wonderful" sound of a "no feedback" amplifier is the wonderful sound of lots of instabilities and underdamped oscillations. You may like it, I don't, it isn't music.

Thus, those that choose to build stable, repeatable, and rationally priced amplifiers will use some feedback. The catch is in knowing what the feedback can and cannot do in the real world, and to use the feedback properly, so that the unit does not only measure well, but actually performs well under transient conditions in the real world.

Now, back to that vacuum tube amplifier. Remember we mentioned that if the voltage amplifier and the current amplifier were perfect, everything would be just fine. Sorry folks, the internal circuits are not perfect and that is where the troubles start.

Inasmuch as the feedback is supposed to compensate for any non-linearities between input and output, it is nice to know what non-linearities exist.

First of all each individual active device is non-linear. Its transfer characteristics are exponential, not linear, be it tube, FET, or transistor. Refer to the sketches below. If the device was perfect, its transfer function would be a straight line, and the slope would remain the same at all frequencies.

The actual characteristics are shown in the second and third sketches. Note that the characteristics are actually exponential. The device is only very linear near the center-line of its operation, and the harder it is "worked", the less linear it becomes, finally becoming 100% non-linear when its absolute limitations are reached.

In addition, the slope of the transfer function becomes less at higher frequencies as the gain of the device reduces.

Thus if one attempts to get the same output from any given device at higher frequency, one will drive it into gross non-linearity sooner, as its headroom is less.

In a similar fashion, at very low frequencies (approaching DC) the device becomes more non-linear and its gain drops. In addition, because the slope changes with frequency, a kind of phase distortion is introduced which is not measured in standard IHF distortion tests, which measure only single frequency performance.

Obviously, to optimize internal linearity, it is very desirable to operate each device within as narrow a bandwidth and as limited an amplitude range as possible while still covering the audio frequency range of interest.

In a vacuum tube amplifier another major non-linearity is the output transformer. The output transformer's primary coil is just that — a very large coil (inductor) in series with the output tubes. Obviously the coil becomes very resistive as the frequency goes up, and in an audio amplifier this happens well within the audio range, rolling off the high frequency output. At very low frequencies the core of the transformer saturates giving very non-linear bass performance. If one wants good high frequency performance then one must have very small output transformers so that the coil inductance is low. If one wants good low frequency performance one must have very large transformers so the core does not saturate. These requirements are mutually self exclusive. These requirements become more difficult to meet as the power rating goes up. If one attempts to "get around" this by designing a tube amplifier without output transformers, then one is faced with the problem that output tubes have very high output impedance and will not drive normal loudspeaker loads (8 ohms nominal) without severe non-linearities.

Since the output transformer is a very narrow band and non-linear device, it is obviously necessary to not feed into the transformer a signal that it cannot handle — the bandwidth of the amplifier must be limited to within the bandwidth of the output transformer. You cannot stuff 10 pounds in a 1 pound sack.

The power supply is another source of distortion. The power supply can be considered to be in series with and part of the output circuit. All current that flows through the output circuit and speaker load first must flow through the power supply. The frequency limits of a power supply are real. It can be modeled as an inductor in series with a capacitor. Obviously at DC the capacitor's impedance is infinitely high, and at high frequencies the inductance is infinitely high.

Thus, at very low and very high frequencies, the power supply is not capable at all. Unless great care is used in the power supply design, it may have multiple resonances, and actually be high impedance at many frequencies within the audio range. Consider also that since the power supply is part of the output circuit, if somebody offers a “wonderful” vacuum tube amplifier with a “wonderful” solid state power supply, you no longer have a vacuum tube amplifier, but a solid state amplifier, so how can it be a “wonderful” tube amplifier?

Of course the power supply is also attached to the voltage amplifier section. Consider that all current draw by the output section causes a signal to show up on the power supply feeds. Any given device or circuit will work best when its supply is absolutely stable. Circuits are designed to reject power supply variations, but the supply rejection isn’t absolute. Thus the more signal that shows up on the power supply feed to the voltage amplifier, the more distortion and instabilities there will be, as this is a signal injected into the circuit at the wrong place. Since we already know that the power supply is less effective at very high and very low frequencies, obviously the power supply related distortions will be greater at very high and very low frequencies. Again, a very good reason to bandwidth limit the amplifier to within the capability of its power supply.

Understand of course that we are considering basic ground rules in general. There are many different kinds of voltage amplifier and current amplifier configurations that work fairly well, some simple, some complex. The important thing to know is that unless the circuits are executed to obey the guidelines established above the distortion will be very high under real world conditions, no matter what the linearity of each section is and no matter how high a quality of parts are used.

Another common problem with vacuum tube amplifiers is the value chosen for the interstage coupling capacitors. In the case of the Dynaco St-70 for example (see attached schematic) coupling capacitors C10 and C11 are .1 μF. This introduces another large low frequency roll-off within the feedback loop. Since the amplifier actually amplifies the difference between input and feedback, and since the feedback is the difference between input and feedback, and since the feedback is taken off the output of the amplifier, at low frequencies the difference signal becomes very large partially due to the roll-off caused by the .1 μF coupling capacitors. Now remember that we have shown that any circuit becomes less linear with increasing amplitude and at the frequency extremes. The roll-off caused by the small value interstage coupler makes the front end work very hard to generate a large low frequency correction signal. This causes the front end to run in a very non-linear mode at low frequencies. You hear it as “muddy bass.” The “cure” is quite simple, make the interstage capacitor large enough in value so that the loop roll-off is minimized, thus reducing the correction required, and letting the front end run in a more linear mode. The low frequency correction signal is easy to see on an oscilloscope. Using a low frequency square wave as a source (20 Hz is fine) look at the signal on the output side of the interstage coupler. Note that it looks much like the input signal. Now look at the signal on the input side of the coupler. You will find the circuit is generating a signal with a large bass boost! (This is true in most tube preamps too!) What is happening is that the “flat” input signal is rolled off by the interstage coupling capacitor. Then the rolled off signal is fed back to the input and a correction signal is generated with a large bass boost to make up for the roll-off. The boosted signal is then rolled off again by the coupling capacitor and its output looks just fine. But the “monkey motion” has ruined the voltage amplifier’s linearity at low frequencies.

Now lets look at what is wrong with the original Dyna St-70 in detail. Refer to the audio channel schematic again, keeping in mind that the “dashed” section is our addition, the original has the input connected to V2 directly with a piece of wire.

What we have is a typical vacuum tube amplifier with unlimited bandwidth input acceptance (DC coupled) but with limited bandwidth output transformers and small interstage coupling capacitors. The power supply is also limited bandwidth, being pretty feeble at both low and high frequencies.

A very low frequency signal is rolled off by the interstage coupling capacitors, turned into a “lump” as the output transformer core saturates, and is further distorted as the power supply runs out of steam. The feedback signal, being taken off after all the disasters occur is very different from the input signal. This generates an enormous “difference” signal which drives the front end into 100% distortion trying to make an impossible correction. Inasmuch as the circuits are underdamped too, the “blob” makes the amp ring for a few cycles attempting to digest the mess. Some people call these distortions and ringing, which extends up into the mid-range, “concert hall sound.” Sorry, it isn’t concert hall sound, it is distortion. If you like it you have bad taste.

At high frequencies the compensation in the voltage amplifier rolls off the signal, the active devices roll it off further, and the output transformers attenuate the highs further yet. This generates another huge correction signal at high frequencies, again more than the headroom of the front end, clipping the correction signal once again. Of course the high impedance of the supply has further compounded the problems. The amp is driven into hard slew limiting and all signal entering the amp while any internal device is slewing is erased. Gobs of high frequency distortion are added and part of the music is forever lost. It is very strange to think that some people use the St-70 to drive tweeters when it doesn’t “tweet” at all — it does kind of “squeak.”

Obviously the power supply of the St-70 must be much improved. NO! Not necessarily! Think a minute. Consider that the power bandwidth of the power supply must be greater than the bandwidth of the audio circuit. There are two ways to get this ratio in proper order. The expensive (and stupid) way is to build a huge power supply — and if the amp has DC coupled inputs you can never make it big enough. The easy and smart way is to limit the bandwidth of the circuit to within the capabilities of the existing power supply, especially if it is absolutely necessary to bandwidth limit the inputs anyway to make the input bandwidth within the capabilities of the output transformers.

As mentioned earlier, the interstage coupling capacitors, C10 and C11 are too small. Note that as long as the input is DC coupled, it is not possible to make C10 and C11 big enough, as even a very large capacitor will have an inside the loop roll off when compared to DC input acceptance.

Upgrading the Dyna St-70 Amplifier

To install the input bandwidth limited filter on the St-70 you will need 8 parts: (2) 10,000 ohm resistors, (2) 470,000 ohm resistors (5% carbon film 1/4 watt parts from Radio Shack are just fine, and it would be better if you could use a meter and “pair” them, so they are matched within 1%.) You will also need (2) 1000 pF capacitors (mica, polystyrene, or mylar are OK, of about 100 volt rating - the capacitors used should be physically small) and (2) .02 μF capacitors (film) 100 volt rating, again physically as small as possible. Again, Radio Shack will have adequate parts and if you can match them on a precision capacitance meter it will be helpful. The capacitor values suggested are not absolute. Anything from about .02 to .033 μF is OK for the larger cap, and 800 to 1200 pF for the smaller capacitor.

The new 6 dB per octave low pass and high pass filter is installed on the input jacks on the bottom inside of the chassis. The mono-stereo switch wiring is eliminated as the performance is poorer when bridged mono because of the difference between the two channels (no two output tubes or output transformers are identical) and because the input filter does not work properly in the mono mode.

To eliminate the mono-stereo switch do the following:

1. Remove all the wires from the input jack and mono-stereo switch except for the two wires going directly from the input jack ground lugs to the PC card (these are actually extensions of the leads of two 10 ohm resistors mounted on the card). These remain. Also remove the two original 470,000 ohm resistors from the jack and switch.

2. Remove the two wires running from the hot lugs of the input jack to eyelets 7 and 17 on the PC card.

3. Connect a 10,000 ohm resistor in series with a .02 μF capacitor and connect the capacitor end of the series set to the left channel hot input jack and the resistor end to eyelet 7 on the PC card.

4. Connect another 10,000 ohm resistor in series with a .02 μF capacitor and connect the capacitor end to the right channel hot input jack and the resistor end to eyelet 17 on the PC card.

5. Connect a 1000 pF capacitor in parallel with a 470,000 ohm resistor and install the resistor between the left channel ground lug and eyelet 7 on the PC card.

6. Connect another 1000 pF capacitor in parallel with a 470,000 ohm resistor and connect the resistor from the right channel ground lug on the input jack to eyelet 17 on the PC card.

Now that the input bandwidth is set to a rational, finite limit, it is possible to make the interstage coupling capacitors "big enough." You will need to acquire four 1 μF at 400 volt film capacitors (mylar, polypropylene, or whatever). Again Radio Shack will have adequate parts.

Locate and remove the four large identical black tubular 0.1 μF at 400 volt capacitors from the PC card. They are positioned parallel with the front of the chassis, one at each corner of the PC card.

Replace each with a 1.0 μF at 400 volt capacitor. The exact value of the replacements is not critical. They can be anything between 0.8 μF to 2.0 μF at 400 volts or higher. It is important that all four new capacitors be exactly the same.

Further detail improvements can be made to the St-70 The bias supply capacitors in old St-70 amps should be replaced. We suggest that the two original 50 μF capacitors (C3 and C4, located on the 7 lug terminal strip under the chassis) be replaced with new 100 μF at 80 volt electrolytic capacitors (again, available at Radio Shack). Note that the positive end of each cap is connected to ground. Do not use a larger capacitor in this application or the supply will come up too slowly, over-biasing the output tubes at turn on.

Although the original power supply is now adequate, further reductions in hum and noise can be made by installing an additional 100 μF at 500 volt electrolytic capacitor (a 450 volt rated cap with a 500+ volt surge rating is adequate unless you have high line voltage) from pin 8 of the power supply tube (V1 — 5AR4) to chassis ground at the ground lug near the quad filter. The positive end of the cap goes to the tube socket, the negative end to ground.

Inasmuch as the perceived "image" and "depth" of an audio system is dependent upon both channels having exactly the same gain and phase response, and because the resistors in the St-70 (and other tube amps) may have drifted out of specification over the years, it is helpful to replace all of the resistors with new tight tolerance parts. The gain determining resistors especially should be matched to each other within l%. The RN60D and RL42S metal film resistors shown on the attached St-70 parts list are a good choice. However using 1/2 watt carbon film resistors for the RN60s and 2 watt carbon film resistors for the RL42S types is just fine, except you will have to sort more of them to get a tight match, channel to channel.

Because selenium rectifiers (the small little finned block located in the bottom middle of the chassis) become resistive with age, you may be able to increase the voltage to your bias supply by substituting a 1N4004 silicon diode for this part (D1). Because the negative voltage to the bias supply will now be higher than stock, it probably will also be necessary to change the value of R1 (10,000 ohm 2 watt resistor) to 18,000 ohm 2 watt to allow the amp to bias adjust at 1.56 volts DC across R20 in the center of rotation of P1 and P2.

The St-70 and other tube amplifiers run very hot. This tends to make solder joints deteriorate with time. Re-solder all solder connections in the amplifier, including all parts, leads, and the tube sockets on the PC card. Clean the input jacks, output terminals, the bias pots, and all the tube sockets with Cramolin R-5 Spray (we have 6 oz. spray cans available for $20.00). Usually lightly "crimping" the hot (inner) terminals of the input jacks will eliminate patch cable dropouts.

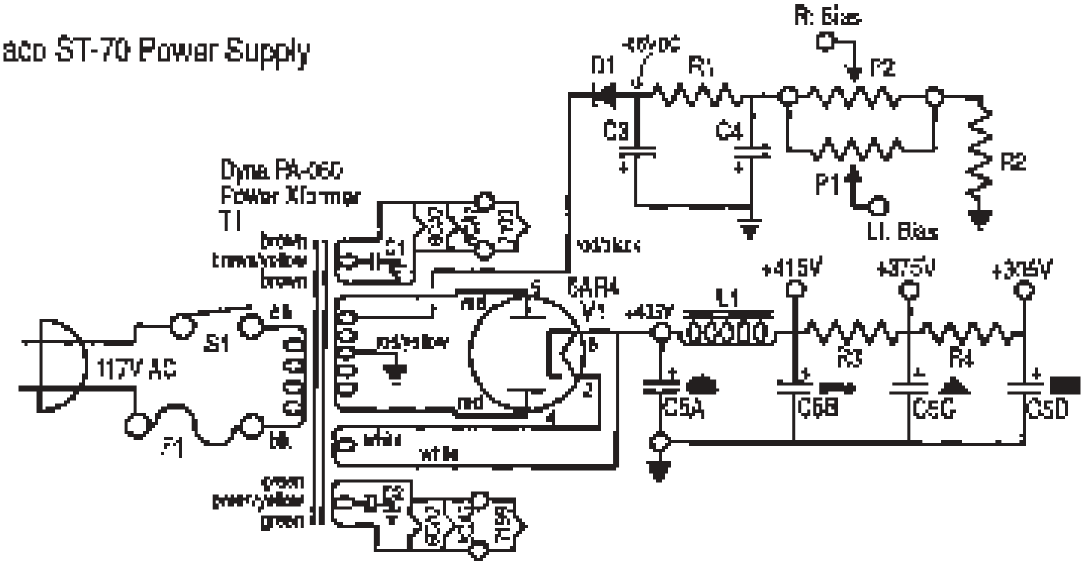
1990 Notes: In the St-70 the noise characteristics, gain, power, and slew rate are dependent upon having high quality tubes in the unit. We have had good luck with the complete $90.00 tube set that Sound Value/Stereo Cost Cutters sells. The set consists of four National 6CA7 tubes, two Philips ECG 7199 tubes and a Philips 5AR4 rectifier. We do not sell vacuum tubes except for the JoLida 12AX7A tubes we use in the SUPER PAS THREE preamplifier. Refer to the attached schematic and parts list for other service and adjustment notes on the St-70.

I assume you have noticed we have not spent much time on the inner details of the circuit topography of the St-70. There may, or may not be "better" input, phase inverter, and output circuits available. The point is that almost all tube amps are mistakenly DC coupled and whatever the internal circuits are, they are driven into gross non-linearities. The important concept is that any tube amp in which the input is limited to within the internal capabilities of the circuit will outperform any tube amp that can be driven into internal overload, no matter how expensive or complex the circuits may be. And the final limitations of a tube amplifier are the output transformers. Lots of money spent trying to achieve a "better" drive circuit probably is of little value, because the output transformers still are the limits of performance.

Things You Should Not Do To Your Vacuum Tube Amplifier and Why.

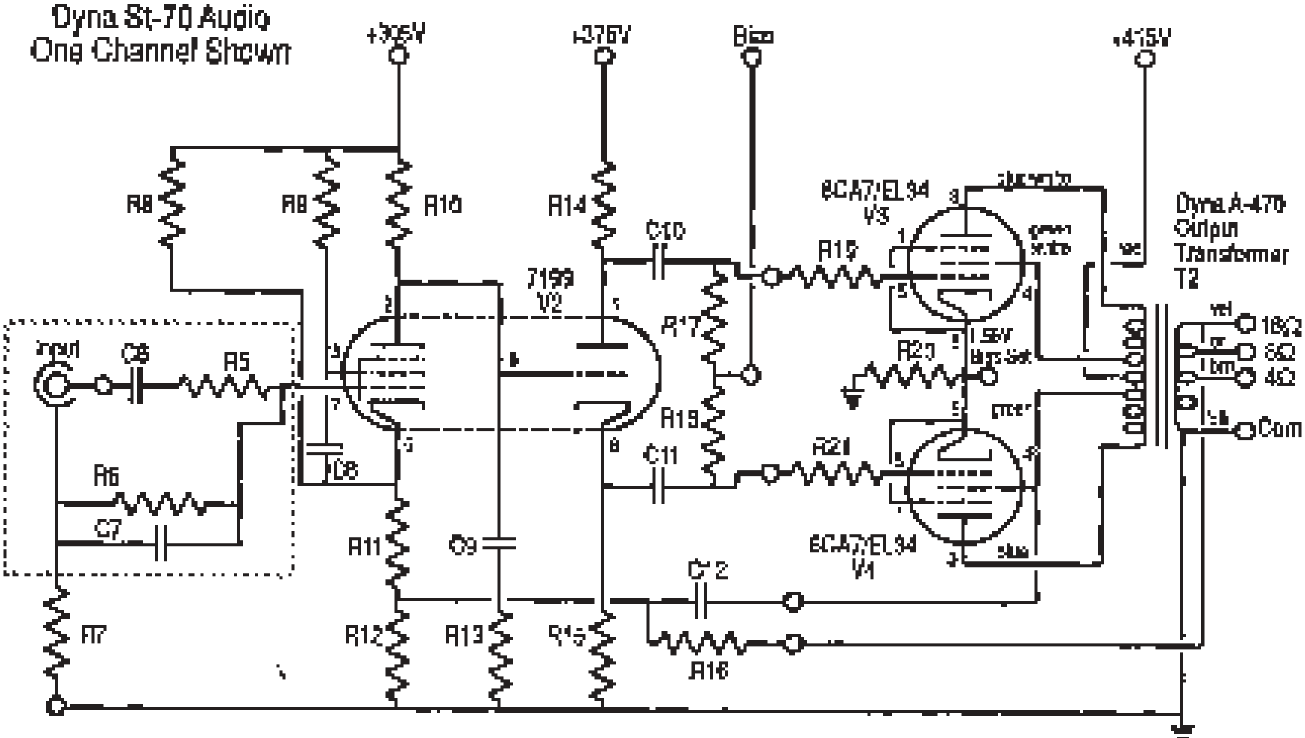
DO NOT install a solid state diode bridge to replace the vacuum tube rectifier. The supply is

Dynaco ST-70 Power Supply



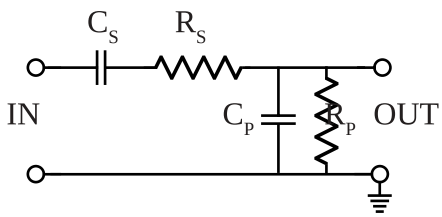
Suggested 1982 modifications to Dyna St-70 Amplifier by Audio by Van Alstine

Dyna St-70 Audio
One Channel Shown



Note: Input circuit (C6, R5, C7, and R6) is new and replaces the original direct coupled input circuit. All wiring is removed from the input jacks except R7 leads and then the new input circuit is installed. The stereo-mono switch should be eliminated. C3, C4, C10, and C11 are revised values. Adjust bias (P1 & P2) for 1.56 volts across R20. L1 can be replaced with a 30 ohm 10 watt resistor to make amplifier operational if L1 is defective. This input filter circuit is also helpful for the Dyna MKIII and MKIV amplifiers.

Formula to Solve 6 dB/Octave High & Low Pass Filter Such as Recommended for the ST-70



C_s = seriesinput capacitorin Farads
R_s = seriesinput resistorin Ohms
C_p = parallelinput capacitorin Farads
R_p = parallelinput resistorin Ohms
$f_{low\,(-3dB)}$ = low frequency cutoff frequency in Hertz
$f_{high\,(-3dB)}$ = high frequency cutoff frequency in Hertz

$$a = R_sC_sR_pC_p$$

$$b=R_sC_s+R_pC_p+R_pC_s$$

$$f_{low\,(-3\,dB)} = \frac{b - \sqrt{b^2-4a}}{4\pi a}$$

$$f_{high\,(-3\,dB)} = \frac{b + \sqrt{b^2-4a}}{4\pi a}$$

Note: These formula assume the driving source has a reasonably low output impedance and that it is driving into a reasonably high input impedance.

ST-70 Input Filter Example

All resistors must be converted into Ohms and all capacitors must be converted into Farads:

$$R[\text{Ohms}] = R[\text{K Ohms}] \times 10^3$$

$$R[\text{Ohms}] = R[\text{M Ohms}] \times 10^6$$

$$C[\text{Farads}] = C[\mu\text{Farads}] \times 10^{-6}$$

$$C[\text{Farads}] = C[\eta\text{Farads}] \times 10^{-9}$$

$$C[\text{Farads}] = C[\rho\text{Farads}] \times 10^{-12}$$

$$C_s = C_6 = .02 \times 10^{-6} \text{ Farads}$$

$$R_s = R_5 = 10 \times 10^3 \text{ Ohms}$$

$$C_p = C_7 = 1000 \times 10^{-12} \text{ Farads}$$

$$R_p = R_6 = 475 \times 10^3 \text{ Ohms}$$

$$a = R_sC_sR_pC_p = \left(10 \times 10^3\right)\left(.02 \times 10^{-6}\right)\left(475 \times 10^3\right)\left(1000 \times 10^{-12}\right) = 9.5 \times 10^{-8}$$

$$b = R_sC_s + R_pC_p + R_pC_s = \left(10 \times 10^3\right)\left(.02 \times 10^{-6}\right) + \left(475 \times 10^3\right)\left(1000 \times 10^{-12}\right) + \left(475 \times 10^3\right)\left(.02 \times 10^{-6}\right) = 1.0175 \times 10^{-2}$$

$$f_{low(-3dB)} = \frac{b - \sqrt{b^2-4a}}{4\pi a} = \frac{1.0175 \times 10^{-2} - \sqrt{\left(1.0175 \times 10^{-2}\right)^2 - 4 \times 9.5 \times 10^{-8}}}{4 \times 3.14159 \times 9.5 \times 10^{-8}} = 15.656 \text{ Hertz}$$

$$f_{high(-3dB)} = \frac{b + \sqrt{b^2-4a}}{4\pi a} = \frac{1.0175 \times 10^{-2} + \sqrt{\left(1.0175 \times 10^{-2}\right)^2 - 4 \times 9.5 \times 10^{-8}}}{4 \times 3.14159 \times 9.5 \times 10^{-8}} = 17030 \text{ Hertz}$$

Also Note: These results do apply since a typical preamp has reasonably low output impedance and since the input impedance of the 7199 tube in the St-70 is reasonably high.

operating at 500 volts with line surges over 1000 volts! There are no reliable diodes available to operate at this voltage. You will be in great danger of blowing the diode bridge and damaging your power transformer and filter capacitor. In addition, the solid state supply will "turn on" instantly, and the full B+ voltage will be fed to the tubes before the heaters have warmed up and turned the tubes on. This will tend to over voltage the quad filter capacitor and capacitors downstream, which may damage them. The output tubes will run hotter than normal and have a short service life. There are no useful redeeming advantages to a solid state diode bridge.

DO NOT install solid state regulators. The "aftermarket" circuits we have seen use transistors with inadequate voltage ratings (operating in the "blow-up" mode) and have severe slew rate limitations. Remember, the bandwidth of your power supply must be greater than the audio circuits, and a series bi-polar regulator is bog slow! It will change the sound, it makes it much worse!

DO NOT add external power capacitors. The long hookup wires will have lots of inductance and impair the high frequency performance.

DO NOT rewire the amplifier internally with "magic wire." The chances are you will screw up the lead routings, add longer lead runs than the original and increase stray inductances. The probabilities of internal short circuits and bad connections increase as the wires are too large for reliable termination.

DO NOT replace your capacitors with high priced and physically large "wonder caps." The larger the physical size of a given value capacitor, the greater its inductance will be, and the more trash it will dump into the circuit. Magic "wonder caps" do change the sound, they make it worse!

DO NOT use polystyrene capacitors near heat generating components. They change value with temperature, and near an output tube they may even melt.

DO NOT ship your vacuum tube amplifier to us to fix if you screw it up. Output tubes don't survive shipping, tube amps are heavy and expensive to ship, and their performance is limited. One of the "joys" of owning a vacuum tube amplifier is learning how to fix it yourself. If you don't want to do this, you shouldn't own a vacuum tube amplifier.

DO NOT hesitate to call us if you have any questions. If you send us the schematic for your amplifier we may be able to give you detailed advice, but expect to pay if the work requires the services of my electrical engineer. NOTE: Stereo Cost Cutters, Box 551, Dublin, Ohio 43017 has many repair parts available for Dyna tube equipment. Their phone number is 614-889-2117.

Parts for St-70 Upgrade

C1	0.02 μF 1000V disc
C2	0.02 μF 1000V disc
C3	100 μF 80V electrolytic
C4	100 μF 80V electrolytic
C5A	30 μF 525V electrolytic
C5B	20 μF 525V electrolytic
C5C	20 μF 525V electrolytic
C5D	20 μF 525V electrolytic
C6	.02 μF 50V film
C7	1000 pF 50V film
C8	.05 μF 400V film
C9	82 pF 500V mica
C10	1 μF 400V film
C11	1 μF 400V film
C12	390 pF 500V mica
D1	1N4003 or 1N4004 silicon diode
V1	5AR4 rectifier tube
V2	7199 pentode, triode tube
V3	6CA7/EL34 pentode tube
V4	6CA7/EL34 pentode tube
P1	10 KΩ bias trimpot
P2	10 KΩ bias trimpot
R1	10 KΩ 2W
R2	10 KΩ 2W
R3	6.8 KΩ 2W
R4	22 KΩ 2W
R5	10 KΩ 0.5W
R6	475 KΩ 0.5W
R7	10 Ω 0.5W
R8	330 KΩ 2W
R9	1.5 MΩ 0.5W
R10	270 KΩ 2W
R11	620 Ω 0.5W
R12	47 Ω 0.5W
R13	18 KΩ 0.5W
R14	47 KΩ 2W matched within 1% of R15
R15	47 KΩ 2W matched with 1% of R14
R16	1 KΩ 2W
R17	270 KΩ 0.5W matched within 1% of R18
R18	270 KΩ 0.5W matched within 1% of R17
R19	1KΩ 0.5W
R20	15.6 Ω 1W
R21	1 KΩ 0.5W
L1	choke, Dynaco C-354
S1	power switch, SPST
F1	fuse, 3 ampere slo-blo 3AG
T1	power transformer, Dynaco PA-060
T2	output transformer, Dynaco A-470

Service Notes

Examining the 1.56 volt bias set test point on the St-70 can tell you much about the condition of the amplifier. If the output tubes are old, it may be impossible to adjust the bias pots to bring the voltage up high enough. Replace the output tubes. A shorted output tube may cause the bias reading to run-away high. 7199 tubes are best selected by examining the output of the amp on a scope. A low gain or noisy 7199 will show excess output hum and/or not make full power. Fuse blowing can be caused by two problems. A hard blow soon after turn-on indicates a power supply short, either a defective 5AR4 tube or a shorted quad filter cap. A fuse that blows soft or after a few minutes of operation indicates a problem with the audio circuits - probably a bad 6CA7 tube. Note that the bias setting will vary with AC line voltage so the value isn't an absolute. It is possible to swap tubes channel to channel (except the 5AR4) one at a time to locate a defective tube.

The new input filter circuit provides -3 dB poles at 16 Hz and 17 kHz which keeps the audio circuit working within the limits of the output transformers. The larger C10 and C11 effectively takes them out of circuit for AC signal purposes after the input filters have been installed. It is much more important to install the new input filter circuits than to replace all of the resistors and capacitors. Match R and C values channel to channel for a good gain match between the channels.

Frank Van Alstine

VOLUME ONE NUMBER EIGHT AUGUST, 1982

I am starting off this month by recommending that you purchase a very interesting and informative little booklet (50 pages) called *HOW TO BEAT THE STEREO RIP-OFF* by Barnet Feingold.

The price is $4.95 + $1.00 postage and handling and it is available from: Geronimo Press, 306 North Plum Street, Northfield, Minnesota, 55057. Minnesota residents add sales tax. [*1990 note: Refer to Barnet Feingold, 8 Bittersweet Road, Fairport, NY 14450 if you want a copy of this booklet now*].

The booklet explains in detail how retail hi-fi stores operate, and the pressures on them (they must make a profit to keep the doors open) that may cause conflicts between their profitability and supplying you the best equipment. It gives excellent insights on how to evaluate equipment on a consistent basis. Chapter headings such as "How to tell whether a shop is worth dealing with," "Undergrounders useful as buying guides," "The pros and cons of mail-order shopping," indicate the thoughtfulness of this book.

It is not another subjective attempt to evaluate specific equipment. It is an outstanding effort at teaching you the skills necessary for you to evaluate equipment (and the shops selling the equipment) yourself.

I consider it a must for a beginning hi-fi shopper and even the experienced audiophile will learn a few things. Certainly it would make a very worthwhile present for another audio enthusiast. Order *HOW TO BEAT THE STEREO RIP-OFF* now from Geronimo Press. Do not order the book from us, we have no financial interest in it. Although we could forward orders, it would only waste time. The book is a work of love by Mr. Feingold. You will like it.

The "meat" of this issue is a discussion of audio interconnect cables and speaker wires from a rational and objective viewpoint. Our goal is to allow you to spend the least money possible for cables that meet a series of rational constraints. We will also point out some of the inconsistencies in the logic used by the arm waving golden ears in subjectively evaluating cables and writing about what cable is best. We will attempt to establish a crude "flow chart" describing the constraints which will allow you to make go/no-go decisions. Do not be put off if the early constraints seem overly simple or obvious. We have seen many so called high end systems where even the most obvious no-no's were violated.

Constraint #1. Is the cable long enough to reach reliably between the equipment it connects? This is not a facetious question. One golden ear reviewer I have visited uses special "super-magic" custom made interconnect cables. They just barely reach (with a little stretching) between his preamp and power amp locations. Only a little bump to the equipment or the shelf causes the cables to pop out of the jacks, causing an open ground, causing fuse blowing hum. Sorry, that does not sound good. In addition the jacks on the equipment are deformed and damaged by the stress. I have watched this reviewer take an agonizing 30 minutes just to change preamps in his system, positioning it "just so" so the interconnects do not fall out. This makes for very inconsistent A-B tests of various preamps. I have also seen runs of very expensive speaker wires spliced with typical amateur gob-soldered connections to make them long enough. It makes you wonder. If the cables are too short, don't buy them.

Constraint #2. Is the cable reasonably reliable for the intended use? I am highly suspect of speaker cables made of woven or braided multi-stranded wires, each only insulated by a lacquer coating. If the very fragile insulation coating on any two internal strands chips or cracks, you risk a dead short across your amplifier output, which, in many cases, will cause expensive damage to the amplifier. Again, blown amplifiers don't sound good. If it appears that stepping on the cable, bending it double, or stretching the cable will cause it to short out, don't use it. One of our audiophile clients purchased new "Cobra cables" to use with his Dyna 416 and KEF 105's. Unknown to him one of the cables had an internal short. The amp promptly blew up. He then tested the cables by reversing them to the opposite channel of the amp. The other channel then blew up. He then sent us the amp to repair. You guessed it, he didn't tell us he had changed speaker cables. You guessed it again. He blew it up again. Then he told us he had new "magic cables." Magic speaker cables may be more expensive than you think!

Constraint #3. Can the cables be terminated reliably? If the speaker wires are so large that they cannot fit into the commonly found terminals on speakers and amplifiers then the cables are worthless. We see no value in super heavy gauge wires if 90% of the wire must be cut away to fit the ends into terminals. When visiting "hi-end" dealers around the country I usually see two kinds of terminations on super heavy gauge speaker wires: 1. Copper fuzz ball terminations – where most of the strands are wadded up in a big fuzz ball, with the actual connection being made by a couple of dangling threads. The fuzz balls being about 1/64" away from shorting out. 2. Battery terminal connections – the speaker wires ending in huge spade lugs each about 1" across. These are about 4 times too large to fit any amp or speaker terminal and again are usually hanging by their very edge and within 1/64" of shorting together. We also see oversized RCA phono plugs so large in external diameter that they will not fit into standard chassis jacks at all, that short together, and that break chassis jacks. The rationale for the use of grossly oversized cables seems to be similar to deciding that since farm tractor tires give excellent traction on farm tractors, they would really be wonderful installed on a Volkswagon.

Constraint #4. Is the cable flexible enough to be reliable? Many magic cables are so stiff and heavy that they tend to pop out of equipment terminals with the slightest jar. Again, open ground hum loops and shorted amplifier outputs do not sound good. (The heaviest speaker wire that I have seen that satisfies constraints 1-4 is standard Monster Cable. To their credit, Monster Cable does supply a variety of terminations for their cable that will make a reliable connection to the equipment you use.)

Constraint #5. Where does the magic in magic cables come from? Do you really want to pay for it?

1. The capital investment in machinery to actually make wires is very expensive and the equipment is huge. Belden, Columbia, and other actual cable makers have plants the size of automobile assembly plants.

   Obviously, few little magic speaker wire makers can afford to own a cable manufacturing plant. Consider then that the little magic wire maker does not make their own magic wire. They buy it from some wire manufacturing plant.

2. The cost to tool up to produce a specific kind of cable is also beyond the means of most little magic wire suppliers. Can you imagine a magic wire supplier going into the engineering offices at Belden and asking "Oh please kind sirs, will you make me 10,000' of a 1700 strand, quadruple twisted, triple braided, Teflon insulated, titanium coated, neon filled wire?" You know what Belden is going to say – "sure, the tooling charge is $200,000.00 and the minimum order is 50 miles."

   Guess what folks, in general, the little magic wire supplier cannot afford to have a special wire tooled up by a wire making plant.

3. Consider then that the "magic cable" is actually purchased from a cable type already produced and stocked by some wire manufacturer.

   Consider then that the cable in question was tooled and produced by the wire manufacturer for some specific purpose and for some good engineering reason.

   Consider that the purpose that Belden and others tool and produce cable is not to have magic sonic qualities.

4. Thus folks, the magic you pay for when you buy magic wires was not engineered into it in the tooling stage.

   The magic was not even produced in the finished product.

   The magic was added on by the magic wire supplier after the fact, by running around in circles, waving arms a lot, and writing unsubstantiated purple prose advertising claims.

5. I submit that after the fact magic is the most expensive kind of magic you can buy, assuming that you care to buy magic at all. I also suggest that magic is more expensive than electrical engineering.

6. For example, we have recently seen a magic interconnect cable claiming to have an "air core" with the magic qualities of being "very quiet." The price, being magic, is about $10.00 a foot. A little research indicates that the actual cable used is similar to Belden #8254 93 ohm semi-solid polyethylene coaxial cable. Belden's only claims are that the nominal velocity of propagation is 84% (normal) and that the attenuation is about 3.1 dB per 100' at 100 MHz (normal). Belden's price for the cable is about 15¢ per foot in 1000 foot rolls. Although we realize that labor is involved to cut and strip the cable into interconnect lengths, and that RCA phono plugs are furnished and installed, and that the quality of workmanship for this particular finished "magic cable" was very good (unusual), still the final price for the magic is about 100 times the price of the original engineering.

By the way, if you would like to learn more about the real characteristics of a wide variety of cables and wires, we suggest that you get a copy of

Belden's electronic wire and cable catalogue. It has a price of $3.00 and we suspect that an additional $2.00 would cover shipping to you. Write to Belden Corporation, Sales Department, P.0. BOX 1331, Richmond, Indiana 47374.

For example, the Belden catalogue has a little chart (with the mathematics to support it) that shows the appropriate wire gauge to use for your speaker wires. It would indicate, for example, that if you use an 18 gauge wire (similar to Radio Shack speaker wire) you can have up to a 70 foot long run into an eight ohm load with less than 10% power loss (less than 1 dB and inaudible). This means that in the audio range of interest, the effect of this cable, as compared to a zero resistance cable, would be that you may have to turn your volume control up one-half step, and that would be the only effect. I would rather turn my volume control up one-half step than buy a heavier cable.

Another interesting bit of data from the Belden catalogue is found by examining the specifications of a wide variety of coaxial cables. Typically, the high frequency attenuation is about 3 dB per 100 feet of cable at 100 MHz. Thus 3 feet of cable will have less than .03 dB of loss at 100,000,000 Hertz. Consider then that the actual loss at 20,000 Hz (5000 times lower frequency) is so close to zero that it cannot be measured. Thus we suggest you ask yourself the following question before considering the purchase of magic cables claiming "flatter frequency response" or "extended highs": Are my phono cartridge, preamp, amplifier, and speakers absolutely linear to within .000001 dB at 20,000 Hz?

Finally, a real world wire can crudely be modeled as a resistor and inductor in series, with a capacitor and resistor in parallel. The driving source (preamp or power amp output) can be modeled as some kind of combination of R, C, and L, as can the load (amp input or speaker). Thus the actual frequency response of this complex tuned circuit at some high frequency (fortunately well above audio) begins to exhibit multiple resonances. Inasmuch as the characteristics of the resonances change with characteristics of the load, characteristics of the source, and length of the cable, as modified by the location of the cable in its real world surroundings, you are presented with a interesting quandary.

1. Let us assume that golden ear cable reviewer really can hear differences in the sonic quality of cables which are an indirect manifestation of the real world ultrasonic resonances in any cable-load system.
2. Let us assume that golden ear really has accurately sorted out the subjective differences between 80 different brands of cables in his pet system and has indeed found one cable he "likes" best. Note that since all combinations resonate, he cannot find the "best" cable, but only the cable who's real world characteristics are least undesirable in that specific system
3. The quandary is that you still cannot use reviewer's findings. Remember the ultrasonic resonances are created by the real world characteristics of the source, load, cable, and length of cable.

   No two audio units have identical characteristics, not even two samples of the same preamp, power amp, or speaker. It is unlikely that any two lengths of cable are exactly the same length, or are located exactly the same, or even have identical physical properties.

   Thus, golden ear's findings (for whatever they are worth) are valid only for that specific cable, attached to that specific equipment, in that specific room.

   In your room, with your equipment, the resonances in the cable will be different. There is no way that subjective findings about cable "quality" can be transferred to a different system.

We also offer a general caution: The "Emperor's New Clothes" syndrome strikes hard at audiophiles. By this, I mean that almost always, if it is reported that something "sounds" better (be it magic wires or magic capacitors) and the audiophile knows in advance that said magic part is supposed to "sound better", and if audiophile actually installs same in his system, it will always make the system "sound better", even if the real world effect is to make the system less linear. Any change is perceived as a change for the better no matter what the reality of the situation is. We have seen people turn a fine system into an awful mess before finally figuring out that something isn't right.

Constraint #7. You can't win. No matter what expensive magic cable you buy, no matter how carefully you search, no matter how hard you listen, no matter how much you spend or how bizarre the cables are, no matter how far away from you they are made (I understand audio cables from Tibet are best), you can't win. Next week, some golden ear will come up with a magic cable that is vastly better sounding and three times as expensive as the ones you just bought. Once you "believe," you are going to spend lots of money and not get ahead.

Constraint #8. Some magic cables really do change the sound of your system, they make it much worse. Inasmuch as many amplifiers become very non-linear driving capacitive loads, using a high capacitance speaker cable (multi-conductor "braided" type) is pretty stupid, unless you really like the sound of oscillations. Likewise, the current drive capacity of many preamplifiers is very limited (vacuum tube preamps can only dump 3 milliamperes of current into a load). Large capacitive loads require lots of current to charge. Thus it isn't a very wonderful idea to use long and/or highly capacitive interconnect cables on the output of a vacuum tube preamp. Thus the idea of short speaker wires and long interconnect cables may change the sound, it may make your preamp current limit (slew) and make things worse!

Constraint #9. Some golden ears recommend the use of non-shielded wire for interconnects "unless you have RFI in your area." We offer a simple test for RFI: A. Turn on your radio or TV. B. Does it work? C. If so, congratulations, you have RFI in your area. D0 NOT use unshielded interconnect cables.

A few final observations and conclusions:

1. In my own shop, and in my travels around the country to many hi-end shops, never have I found a single individual, including golden ear editors, who can "pick out" the "sound" of an interconnect cable or speaker wire when I made the substitutions without letting anyone know in advance what cables were changed (excepting high capacitance speaker wires which make many amplifiers perform worse).
2. We can find no golden ear claims for cables, be it construction, insulation, purity of materials, whatever, that are supported in the objective engineering data for that kind of cable.
3. There are known major non-linearities in audio equipment. All phono cartridges are pretty terrible, a "good" speaker is one with only 3% distortion, the master tape your records were cut from were distorted, as was the master cutting machine. Audio electronics have documentable large non-linearities too. In comparison, a rational interconnect or speaker wire, in the audio range, performs so close to a "perfect" transmission line that the actual non-linearities approach zero.

Our recommendation: If you spend more for speaker wires and interconnect cables than what they cost at Radio Shack, you are wasting money. But, if you want to do it, be our guest, as long as you don't violate the above constraints and make your system unreliable.

Frank Van Alstine

VOLUME ONE NUMBER NINE SEPTEMBER, 1982

We are going to discuss How to Troubleshoot Your Audio System this month but first, a short revisit and follow-up regarding "magic capacitors."

Rumors have come to us of the great mystical and magic sonic qualities of WIMA "600 volt per microsecond" slew rate German "high speed" polypropylene capacitors. In fact some of the audio hippies out there claim to "modify" Hafler and other equipment using all "600 volt per microsecond" WIMA capacitors. We investigated.

We have obtained detailed engineering specifications and prices for WIMA capacitors. We have also obtained a couple of samples of Hafler DH-200 amplifiers "professionally modified" claiming to use "high speed" WIMA capacitors.

Regarding the modified Hafler DH-200s, each came to us because, after the "professional" modifications, they had blown up like a small A-bomb, and since the PC cards had been butchered beyond recognition, the end user had no possible repair recourse except to send the units to us for our all new internal circuits, as we install new PC cards of our own design, and care not whether we ashcan stock Hafler cards or ones looking like they had served as part of the street barricades in the Russian revolution. In both cases, the PC cards were mangled with resistors standing on end and spliced together, oversized capacitors tacked all over the back of the cards, transistors substituted of too low a voltage rating for reliable operation, a huge input capacitor taped to the heat sink on one amplifier effectively coupled the output of the amp back into the input, making a Hafler DH-200 100 watt oscillator, not desirable, but it did make it sound different.

In addition, the units were rewired with bizarre magic cables, with typical "fuzzball" connections (in one amp there were cold solder joints both at the AC power switch and at the diode bridge). Speaker fuses were removed and the B+ fuses installed ahead of the diode bridge, insuring destroyed output circuits in the case of accidents as then the supply capacitors would discharge thru the output circuits before any protective fuse blows. This however, wasn't what made us mad.

What made us mad was that none of the "modified" Haflers actually had the "600 volt per microsecond" slew rate capacitors they were claimed to have. Yes, WIMA does make some capacitors with a claimed 650 and 700 volt per microsecond slew rate. However, not all WIMA caps have this specification (see below for discussion of the validity of this specification) and none of the WIMA capacitors used in the Hafler mods had anywhere near the claimed specification. Some of the WIMA caps used were 0.1 μF at 160 volt or 250 volt series MKP 10 which WIMA specifies to have a 100 volt per microsecond slew rate, not 600 volt per microsecond. The input couplers were WIMA MKP 10 2.2 μF at 160 volt which has a 50 volt per microsecond slew rate. These are, for your information, metalized polypropylene film capacitors (PC card mount type) and cost about 20¢ each for the 0.1 μF and $l.25 each for the 2.2 μF values. Another Hafler "modification" had WIMA MKC 4 series 4.7 μF at 100 volt metalized polycarbonate input capacitors. These, dear readers, have a 3 volt per microsecond slew rate, not 600! This, I claim, is FRAUD! ! So, does using super wonderful 600 volt per microsecond WIMA high speed capacitors make your equipment sound "wonderful?" Hell if I know, the units I have evaluated don't use the parts they claim to use. The MKC-4 series 4.7 μF cap costs $l.08 each, for your information. You may be buying very overpriced "sugar pills."

Regarding "slew rate" specifications for capacitors, there is no industry standard for this rating. WIMA, for example, specifies this rating is for a 1.6 volt input for 2 seconds. They do not specify what happens when the rating is exceeded. Do they blow up, distort (how much?), become resistive, become inductive, change value, or whatever? Who knows? Thus you cannot compare "slew rate" rating from one brand of capacitor to another because different manufacturers use different standards. A "fast" rating from one supplier might be a "very slow" rating from another manufacturer if a different testing standard is used. Do not neurose over the "speed" or "magic qualities" of the capacitors in your unit if they have been selected for the following objective criteria:

1. The capacitor must be the proper voltage rating and value for the circuit application. Far too often we have seen 250 volt rated capacitors across 400 volt power supplies, for example.
2. If the capacitor is not "biased on" in the circuit, in a centerline coupling application for example, it must be non-polar type.
3. The capacitor must have adequate stability for the application. In a critical tuned circuit, a highly stable mica capacitor may be required. Obviously, for non-critical applications, such as a .01 μF cap across a power switch, a inexpensive ceramic is just fine.
4. All other things being equal, the capacitor should be as physically small as possible for the given application. The predominate factor determining the inductance of a capacitor is its body size, the inductance being about the same as that of a wire the same length as the body. Large capacitors add more undesirable inductive transient trash to the circuit. This may be the trash some golden ears actually like in describing "magic capacitors."

DO NOT take the above as a complaint against WIMA capacitors. They are just fine and priced competitively with many other similar capacitors. Our argument is with those that misrepresent the properties of the WIMA capacitors, not with the quality or specifications of the product.

Finally, we have recently seen advertisements for "WONDER CAPS" in *Audio* magazine by *International Audio Review*. Inasmuch as all of our legitimate capacitor suppliers will gladly furnish us with engineering data, rational samples, and quantity pricing, and also give us the name and location of the source manufacturer (if the parts are not made 'in house') we have written to IAR requesting this same routine data from them regarding "Wonder Caps." We will let you know next month if IAR responds.

About 3 out of 4 phone calls we get regarding system problems can be traced back to hookup errors. In about half of the cases the end user seems not to have had the information to even decide, with certainty, which component in the system is the cause of the problem. It is most annoying to us to receive the "wrong" piece of equipment to fix when the problem was elsewhere in the system, and I am sure it is annoying to the user too, as taking in the wrong part of the system to get fixed doesn't make his system work again.

Thus it is time to discuss in detail, how to troubleshoot your audio system (no technical skills or tools required).

We will start by making a few basic assumptions. We will assume you have a basic system (phono, tuner, preamp, amp, and a single pair of speakers). We will start first to give methods to determine which component has the problem. We will then go into more detail so you can determine which section of that component has a problem. We will finally add in a few frills, so you can locate problems in crossovers, equalizers, signal processors, bi-amped systems, and more complex systems. If you have vacuum tube equipment, where, in most cases the problems are tube failure itself, you should be able to make most repairs yourself. In solid state equipment that is not user serviceable, at least you should have smaller repair bills when you can tell the service man, "I believe the power supply in my preamp is defective," rather than having to say, "my hi-fi hums, fix it."

Consider first, assuming you have a stereo system, that except for power supply sections and phono stylus assembly, you actually own two complete independent mono systems – a left channel system and a right channel system. Consider also that each system can be considered to be a "chain," starting with the phono cartridge and ending at the speaker. If any link in that chain is broken, the sound will be distorted or missing. Consider also that there are more "links" than the chain of components themselves – that each interconnect cable and its fittings at each end are additional links, and that there are switch contacts inside the equipment that are additional links. All must have electrical and mechanical integrity for the system to operate. That sounds pretty complex, and it is. However, remember that you have two "chains" (the left system and the right system), and as we will describe later, this is handy, as you can substitute "links" from one "chain" to the other to determine exactly where the "break" is.

However, before getting into details, first, do not overlook the obvious! Many, many service calls have been made for problems as simple as these:

1. Are all the power plugs plugged into the AC line? No electricity, no music. It happens all the time.
2. Are all the speaker wires and interconnect cables connected? They get accidently knocked loose, especially stiff, heavy magic cables.
3. Did you check the user replaceable fuses? We have had many amps and speakers to "repair" when the only problem was a blown speaker fuse or amplifier fuse. Replace with the same kind of fuse, never use "slo-blo" fuses for speakers, the speakers will blow first!

   NOTE: Regarding fuses, DO NOT DISCARD BLOWN FUSES if your equipment has a problem. A good serviceman can "read" a blown fuse and save time in troubleshooting the equipment. For example, a speaker fuse that has just "parted" usually indicates a simple "too much power" problem, and unless a driver has been damaged, the cure is a new fuse and don't drive the speaker quite so hard. The other extreme is a fuse blackened with the inside elements splattered all over the glass tube. This means a very heavy overload and probably bung internal parts. Do yourself and your serviceman a favor, save the blown fuses, put them back where they came out and give your serviceman this extra "evidence."
4. Is the tape monitor switch set to "input?" If it is set to "monitor" no source will play except the tape deck playback. We have fixed hundreds of preamps over the years by simply switching the tape monitor switch back to the normal "input" position.

Next, lets try and pin down and define the "strange noises" your system may make (or not make) if it gets sick. It is much easier (and less expensive in the long run) to service your equipment when you can tell the service man, "there is a harsh high frequency hiss from the left channel of my pre-amp," rather than, "my stereo is noisy." You pay for the service time, including the time to find out where the problem really is. You pay less the less time the serviceman has to spend isolating the problem. Let's try and define strange noises so you can give your serviceman better and more definite data.

A. HUM (A predominantly low frequency, rather constant, "hummmmmm" sound) further subdivided as follows:

   1. Rather pure low frequency hum.

      VERY LOUD!

      Almost always an open ground on a interconnect cable. RCA phono jacks lose ground contact (the outer "tulip shaped" contacts) before breaking the hot (inner pin) contact and the resulting "open ground" creates a very loud hum which propagates through the entire system. Turn off system and check all connection cables. Phono input connections and connections from preamp to amplifier are most likely problem areas. A culprit is "after-market" magic interconnect cables which may have out of spec foreign made RCA plugs.

      STEADY, ANNOYING, (but not deafening) HUM.

      If the hum does not change when you turn the volume control up or down (always there even with volume control all the way down, and does not get louder when volume control is turned up) then probably a power supply defect in the power amplifier, most likely a bad power supply capacitor.

      If hum gets louder when volume is turned up, and goes away (or nearly so) when volume is turned all the way down, then hum is probably from one of the source components. Probable causes: Bad phono ground wire connection; phono cartridge interacting with AC field from turntable; turntable located too close to power amplifier; bad power supply in tuner or tape deck; AC power lines located close to and parallel with interconnect cables; hum in source – actually on that particular tape, record, or FM broadcast – not unusual; gremlins - call exorcist!

   2. Hum with additional "buzz" – "ratty" sounding hum. If not resolved by the discussions above may be caused as follows:

      GROUND LOOPS.

      There should be one, and only one, ground connection between each component. In almost all cases, this connection is made thru the ground (outer shield side) of the interconnect cables. Adding an additional wire grounding the chassis together (such as with the third "ground wire" of some "magic interconnect cables") may create a second ground path and a ground loop hum. In some turntable setups it is necessary to disconnect the phono ground wire to minimize hum. Depending on internal power supply decoupling in the components, the orientation of AC power plugs may reduce ground loop hum. Experiment.

      RFI (Radio Frequency Interference).

      A nasty, difficult to resolve "buzzy" hum caused by the equipment or connection cables picking up electromagnetic interference. It may come and go at different times of the day, depending upon when the offending source is operating. A few possible causes are: Poorly shielded interconnect cables (a full braid shield is much better than a spiral wrap); poor contacts (clean all possible mechanical connections with Cramolin Red); actual physical routing of cables can be changed to minimize RFI pickup; use of unshielded magic cables; operation of illegal radio broadcast equipment nearby; operation of a home computer nearby; operation of micro-wave oven; defects in power lines nearby (an arcing distribution transformer can radiate trash over a several block area); use of interconnect cables with a separate ground wire but with only one end of the ground wire connected to avoid ground loops – which makes the ground wire into an antenna; oscillating power supply in DC servo-drive turntable; poorly designed components with excessive bandwidth. In general, well designed equipment with well shielded and maintained interconnections will be reasonably free of RFI problems. Determining where RFI is getting into your system will be discussed next month.

      Another cause of "hum" is because it doesn't know the words.

B. HISS (A predominantly high frequency, rather constant, "hissssss" sound) further subdivided as follows:

   1. A harsh hard hiss, at very high frequencies (combined with a tendency in your system to have tweeters fail and amps blow up for no good reason). Caution! You may have an oscillating component! Unstable amps or preamps may be dumping lots of component damaging ultrasonic trash and all you will hear is a small hard hiss. A common cause is non-engineered "modifications" such as substituting ICs in op-amp preamps that are not stable in the application, adding large "magic" capacitors to PC cards where they couple output to input, improper lead placement inside components. External causes can be having input cables too close to and parallel to output cables, adding "destabilizing" loads such as highly capacitive electrostatic speakers, using "low-inductance" speaker wires with amplifiers designed to be stable only into the normal inductance of a standard speaker wire, old components, originally only marginally stable, that have finally turned into oscillators (old stock Dyna St-120 amps are a good example of a

bad example). Careful! A serviceman may not be able to "see" the oscillations on a hobbyist grade oscilloscope as it may occur at a higher frequency than the operational bandwidth of the scope. Another possibility is impending snake attack.

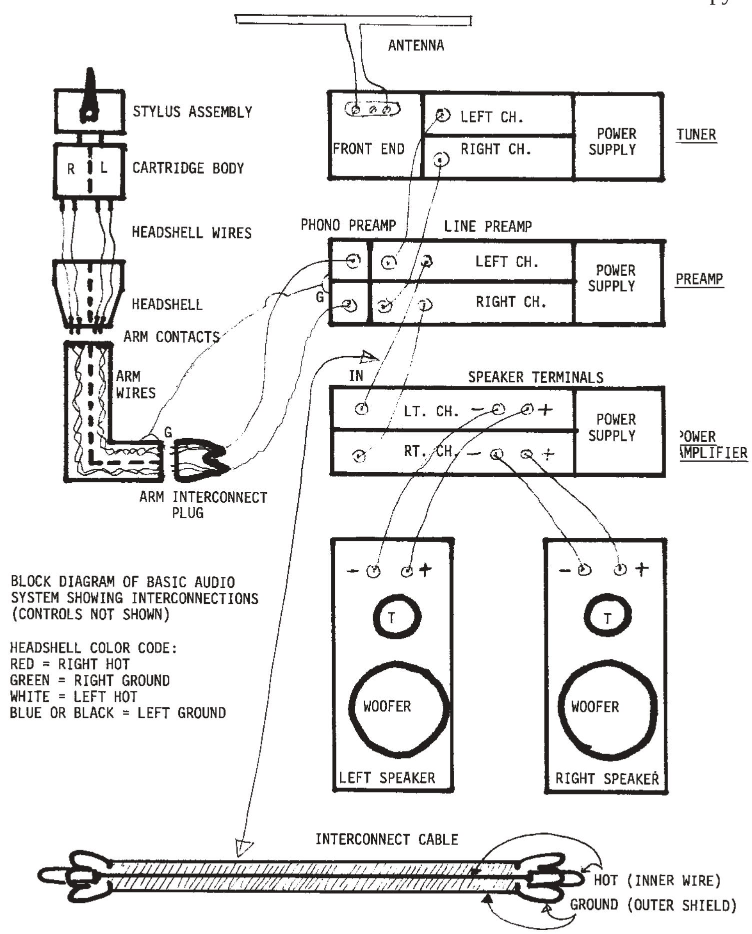
2. Softer, wider-band hiss, sometimes accompanied with a "sputter." This is the sound of a defective (noisy) part in a component. Most likely a noisy IC, noisy tube, followed by noisy resistor, noisy discrete transistor, and least likely, noisy capacitor.

3. Low level, very soft, very smooth hiss (the sound of a distant waterfall). If the level is below that of your records, tapes and FM stations, it is the normal residual noise of a clean audio component.

   NOTE: You can actually use normal hiss to evaluate your system. A good source is interchannel FM tuner noise, in mono, with any mute circuits off. Use the hiss to adjust tone controls and any speaker controls for the smoothest, most natural sounding hiss. Using hiss as a source is an easy way to find defective drivers in a speaker system. It will also help you "cull out" poor speaker systems, as they will sound "peaky." Any tendency toward poor dispersion will show up immediately with hiss as the source of signal to the speaker.

C. SILENCE (no sound at all)

   Refer first to the don't overlook the obvious section in the second column of page 24. If that fails, wait until next month when we will continue with troubleshooting the system.



BLOCK DIAGRAM OF BASIC AUDIO SYSTEM SHOWING INTERCONNECTIONS (CONTROLS NOT SHOWN)

HEADSHELL COLOR CODE:
RED = RIGHT HOT
GREEN = RIGHT GROUND
WHITE = LEFT HOT
BLUE OR BLACK = LEFT GROUND

Next month we will continue, discussing DISTORTION, troubleshooting the system (complete with a few useful sketches), and finally discuss that most vexing problem of all, the "I don't like the way my system sounds today" syndrome. It happens in my system too.

It's about that time of year for those of you with a one year subscription to *Audio Basics* to start thinking about renewing your subscription if you are happy with this newsletter. I can tell you that the $1.00 per issue is not a "break-even" proposition for me. Between first class postage (I am not interested in government subsidies) printing and mailing costs, and the two to three days a month it takes to compose this, it isn't a money maker. I am doing it because I think it needs to be done and nobody else has. However, I will raise the price of *Audio Basics* to $15.00 next year. I will accept subscriptions for next year at the $12.00 price until December 1, 1982. A note with a $15.00 check telling us "subscription renewal" will do. Thanks. [1990 note: The subscription price is now $16.00 per year].

Frank Van Alstine

VOLUME ONE NUMBER TEN OCTOBER, 1982

First of all a follow up on "Wonder Caps." We mentioned last month that we had written to *International Audio Review* asking for engineering data on "Wonder Caps," the same as we would do with any other professional parts supplier. Evidently the request for engineering data (as contrasted to magic wonderful subjective sonic qualities) scared them off; we have had no reply from IAR at all. Any credible supplier will promptly answer requests for engineering data on the parts they supply. Sorry folks, magic claims and non-answers are not credible. We will continue, in the absence of credible engineering information from IAR, to use engineered parts from credible suppliers.

One of our subscribers has informed us that "Wonder Caps" are imported by Reliable Capacitors, Inc., 7409 Bellaire Avenue, North Hollywood, California 45223, phone 213-983-1970. I have of course written to Reliable Capacitors directly, requesting the same engineering data as requested from IAR. Unfortunately, no answer has been received from this company either. Stay tuned for further information if and when we receive it.

Now, back to How to Troubleshoot Your Audio System. Reread *Audio Basics* Number Nine again please, to refresh yourself on the kinds of strange noises your system may produce. To continue, we will mention the final kind of strange noise, and attempt to define it:

D. DISTORTION (Fuzzy, garbled, unbalanced, and or raspy sound)

1. Fuzzy distortion, present only on phono (or tapes made from that record player) probably indicates a dirty stylus assembly or damaged cartridge.

   Acquire a good stylus cleaner such as the Hervic Professional Electronic Stylus Cleaner, the Discwasher stylus cleaner, or the Audio Technicia stylus cleaner and keep your stylus clean. We have had many cartridges returned for "repair" when the only problem was a dirty stylus.

   Examine your stylus for unnatural bends in the cantilever, or a cantilever not centered.

2. Raspy distortion, particularly on heavy bass notes, piano, and other percussive instruments, "in time with" the music is most likely a defective speaker – a voice coil coming apart and rubbing in its magnetic gap, most likely on the woofer or mid-range cone, rarely in the tweeter. The cure is to have the defective component repaired or replaced.

3. Garbled sound is probably an electronic component failure. It can be further subdivided into:

   Garbled output always there, no matter if the system is played loud or soft, and no matter what the source is - phono, tuner, or tape. This is probably an internal defect in the power amplifier or output section of the preamplifier.

   Garbled sound that is only on one source (such as distorted on tuner, but not on phono) is most likely a defect in that particular source.

Garbled distortion that only shows up when the system is driven very hard may just be amplifier clipping or speakers saturating. If this happens with you, and you blow tweeters or other speaker components regularly, you need a much more efficient speaker or much more powerful amplifier, or both.

Garbled sound that seems not to be there at low levels, but shows up when the system is only driven at normal levels is probably the amplifier breaking down under stress and indicates a repair job is in order.

4. Unbalanced sound either in frequency or channel balance can be further subdivided:

A frequency imbalance (no highs, or no bass) is probably a blown driver in your speaker system. If you listen with inter-channel hiss from your FM tuner as a source, you will probably discover that one (or more) drivers in your system have no output at all.

It is also possible that you have your tone control settings way off, such as a problem we once had with a client, now known as the "fat thumb" syndrome. On the stock Dyna PAS preamplifier, the on-off switch is located close to the right channel bass control. We had a client that called often to complain that he either had "no bass" or "too much bass" in his system. When he brought the components out for us to check, we could find nothing at all wrong. It was driving us crazy until I happened to watch him turn the system on in his home. He had big hands, and in the motion of turning on the preamp, was accidently rotating the right bass control full on (too much bass). Of course when he turned the system off, his thumb hit the bass control and rotated it full off (no bass the next time the system was used). Don't laugh, it may be happening to you. Check your control settings each time you use the system.

In many preamplifiers and receivers with tone controls the marked "flat" positions may be far from flat, causing large frequency response errors from channel to channel. The only way to check is to have your unit examined on a test bench and have the tone control knobs set so that the "marked flat" position really is flat. On some older preamps, such as Dyna PAS-3 and Pat-4 units, this may be impossible to do, due to variations in the tone control pots themselves. This is one reason we take the tone controls out of circuit on these units, so that we can get exact channel to channel balance.

Left to right channel imbalance is most likely a phono cartridge problem, assuming the problem occurs on phono only. A phono cartridge will only have equal output from both channels if the cantilever is properly centered in the pole pieces. If your anti-skate adjustment is off, the stylus will be "pulled" to one side (sometimes permanently) and will cause a channel imbalance.

Another culprit is a mistracking volume control. The volume control inside is actually two separate potentiometers on one shaft, mechanically linked together. It is difficult to make two controls exactly the same, and there is always some mechanical "slop" in the linkage. It is not unusual to find 3 dB errors (or more) between the two sections varying with the volume setting. This can cause one channel to have half the output (or less) of the other channel, again varying with the control setting. Many people blame the balance control for the problem, because they have to turn it off of center to get the channels to have equal output. This is an effect, not a cause. Almost always, it is the volume control that is mistracking, so that the balance control has to be adjusted to "make up for the tracking errors" in the volume control.

In the preamps we build, we eliminate the mistracking problems by using Noble stepped precision controls, in which the steps are matched film resistors and the mechanical linkage is very precise. They all track within a small fraction of a dB and give us exact left to right channel balance.

A channel imbalance can also be caused by out of tolerance parts in your equipment. To have the same gain from both sides of your system the gain determining parts must be matched to each other. In one Hafler DH-101 preamp we examined recently, one of the supposedly 2700 pF RIAA capacitors actually measured about 2400 pF. The same part in the other channel measured nearly 2900 pF. This is nearly a 20% difference from one channel to the other and obviously the phono phase-gain balance was adversely affected. It is not unusual for the large 1000 ohm 2 watt gain set resistors in a Dyna St-70 amp to be "cooked" 50% out of specification by heat and age. This too causes a large channel imbalance.

Anyway, if by now you have a better idea of what kind of problem you have, by better identifying the kind of nasty noise your system is making, lets now set up a systematic method of how to find which component, and with luck which part of that component, is causing the problem.

Examine the sketch on this page in which the basic links in your left channel and right channel audio "chain" are identified. Consider that it is possible to swap links from channel to channel to find the broken link.

CAUTION! Make sure the system is turned off before changing any connections. Also make sure all connections are properly made in the first place, (no channels reversed, no speakers out of phase).

First, I will give you an example of a couple of problems, and how they are located.

In the first example, we will assume that the left phono preamp section of your preamplifier has gone bung, a Wonder Cap has shorted out in its circuit and the output of the left phono preamp is shorted to ground (dead). When we turn on the system, what will we hear?

We will have no sound out of the left speaker when, and only when, the system is switched to phono and we try to play a record. At this point many people take their left speaker in for repair, because it "doesn't work on phono." This makes service technician's hair turn prematurely grey. We won't do that, will we?

OK, you have turned on the system and put on you favorite disc, and sob, no sounds from the left speaker. So the first thing to do is to turn the volume up real loud, isn't it, more "power" might make it work again. Right? WRONG!!! Murphy says this is when the channel dropout will cut back in again, just long enough, at full power, to blow your woofer (imported from Yakland) across the living room and imbed it into your spare box of Hungarian vacuum tubes, the channel will shut off again permanently, but not before melting 51.27 meters of Fulton platinum cable and turning your Stasis amplifier circuits into chaos amplifier circuits.

The first thing to do is turn things down and go thru the preliminary checkout listed in Issue Nine of *Audio Basics*. If something smells "funny" or if there is loud hum or a harsh hiss, shut the system off!

In this case a careful inspection finds nothing obviously wrong. All the phono headshell wires are solidly installed, no dirty headshell or arm plug interconnect connections, and nothing seems to be on fire. We try the system again, carefully, and yes, there is no music from the left speaker. Next, we switch the preamp to FM tuner, and hurrah! there is music undistorted from both speakers!

What now do we know? We know the problem is not with the speakers (both work fine), not with the power amplifier (both channels work fine with tuner as a source), not with the tuner (its playing), not with the speaker wires, nor the line or power supply section of the preamp (for it works with a tuner input). We know the interconnects between the preamp and amp, and from tuner to preamp must be fine. Do we know that the phono cartridge works? Nope, not yet, lets find out.

Shut system off, and interchange the left and right phono interconnect cables at the phono inputs on the preamp. Now the left phono channel is fed into the right phono preamp and the right phono channel is fed into the left phono preamp. Turn system on and try a disc again. If the defect was anywhere in the phono system itself (cartridge, headshell wires, arm contacts, internal arm wires, interconnect plug, or interconnect cables between the phono and the preamp) then what would happen? Think about it (???).

Time's up! Obviously, now the left speaker would play, and the right speaker would be dead. Why, because if the "broken link" was in the phono cartridge arm system itself, there would be no output from its left channel, and whichever channel of the preamp phono section you hooked the "dead" input into, would have no output. However, in this particular case, the left speaker is still dead on phono. This does give us some worthwhile information. We now know the phono cartridge and tonearm system and its interconnect connections are just fine. Why? Because the right phono channel worked when plugged into the right phono preamp (the original situation), and now the left phono channel works when plugged into the right phono preamp – thus both phono channels from the record player work.

We now know where the problem is by a process of elimination – the only thing left is the left phono preamplifier section of the preamp, or possibly a dirty switch contact in the selector switch inside of the preamp that selects the output or input of that section. If you have some technical ability, clean the internal switch contacts with Cramolin Red, if its a vacuum tube preamp try a new left phono preamp tube(s), if its an IC preamp with socketed and readily available ICs (many standard types can be found at Radio Shack) try a new IC. Make all changes with the unit unplugged from the AC power line, of course. If all else fails contact the manufacturer for service recommendations. By the way, the manufacturer won't to be too happy to see his perfectly good unit returned chock full of Wonder Caps, so pull them and reinstall the originals (you did save them, didn't you?). In this example, when you try the preamp again, it now works. Do you remember it was a shorted magic cap that was the problem in the first place?

In the second example, we have hum from both speakers (a simple, ordinary hummmm). The hum gets louder when we turn the preamp volume control up further, and gets softer when we turn the preamp volume all the way down, and is there both on phono and tuner, but much louder on phono. Where is the problem? Helpful hint, last week you built a new external regulated power supply for your preamp using special "Valve King" supplied parts kit, which features 300 volt rated capacitors across your 400 volt DC regulator (you don't believe it? – you should see the schematic and parts list a client sent me!). Yep, a blown preamp power supply is the problem. The excess AC ripple dumped into the circuit is causing the hum. The power amp is OK for it doesn't hum as loud when the preamp is turned down (if the hum originated in the power amplifier, it would stay at the same level no matter where the preamp volume was set or which source was selected). The hum is louder on phono because the phono circuits have much higher gain than the line circuits, and amplifies all trash on its supply feed about 20 dB more than the line circuits do. The problem is most likely the power supply because it is common to both channels, and the problem (hum) is common to both channels. It would be very unlikely to have two separate and identical problems occur at the same time in both the left and right audio channels – about as likely as getting two flat tires on your car at the same time from different causes. The most common power supply problems are broken ground wire connections, defective filter capacitors, defective regulator transistors, and in "do it yourself" modifications, improper grounding.

In our third example, the symptom is no sound from the right speaker. This is true both on phono and tuner. When we connect the left channel speaker wires to the right amplifier output, and the right channel speaker wires to the left amplifier output, we then get no sound from the left speaker. We then put the speaker wire connections back as original and again, no sound from the right speaker. We then swap the input connections to the power amplifier (left input interconnect cable to the right amplifier input and right input interconnect cable to the left amplifier input) at the amplifier end only. The problem again changes channels – no sound now from the left speaker. Do we now know that the problem is in the right audio channel of the preamplifier? We recently had a preamp come in "for repair" because of this decision. The owner didn't bother to contact us first for trouble shooting help before returning the unit. Lets look at what we do know after the above procedure.

The problem was a dead right channel. Inasmuch as it occurred both on phono and tuner, we can assume the problem did not originate in either the phono player or FM tuner as the chances of having a right channel defect showing up in two independent sources at the same time is highly unlikely. We could have tested this further by swapping the inputs from one of the sources (say FM tuner), left to right, at the preamp input. If the defect was indeed in the source it would then "change channels." If after the source, it would not change channels. In this case, the problem would not change channels.

When we swapped the speaker leads from the left to right side of the amplifier and the problem changed to the other speaker we then knew the speakers were OK for each played when connected to the left side of the amplifier. When we swapped the interconnect cables at the amp inputs from left to right and the problem again changed channels we then knew that the amplifier was working, as the left amp channel was working originally, and now the right amp channel is working when driven from the left interconnect cable. Thus, both amplifier channels are working.

At this point, we have eliminated the phono system, the FM tuner, the speakers, and the amplifier from being the problem. The only thing that is left is the preamp, and it must go in for service, correct?

Have we overlooked any other possible "do it yourself" tests? Yes, we have! Lets try one more "swap" remembering that the interconnect cables themselves are also links that may be defective. Put everything back as original so that the right speaker is dead. Now lets swap the amp to preamp interconnect cables left to right again, but this time at the preamp end. Now we find that the problem changes channels again, the left speaker is now dead. Interesting! What does this tell us? Nothing yet, but now, lets swap the interconnects left to right at the power amp inputs too, so that the left preamp to amp interconnect cable is completely on the right channel, and the right interconnect cable is completely on the left channel. What happens now? We find that now the problem does not change channels, the left speaker is still dead! We now know that the preamp is working fine! Now the right preamp channel is driving the right amp and speaker channel and it is playing, and originally the left preamp channel was driving the left amp and speaker channel and that was playing. Thus, both channels of the preamp are working. What is left that is causing the problem? The only remaining link (and the broken one) is the right channel preamp to amp interconnect cable (now located on the left channel). Our $99 set of magic Hillbilly Cables (imported from Yakland by the Magic Wire Company at a cost of 39¢ each) have fallen apart internally and the inner (hot wire) is open. If the ground side (outer shield) would have been open we would have had a completely different problem, a very loud hum.

The cure, of course is a quick trip to Radio Shack for a new set of interconnect cables and the system is back in business. Our client who sent in the "defective" preamp mentioned above, got to pay more. He paid $20.00 for a bench checkout, return shipping, and for a new set of interconnect cables. Time to check units that are not defective doesn't come free at any shop, and its kind of difficult to provide free warranty service on equipment that has no problem. In this case a phone call would have given the customer the data to find the bad interconnect cable and saved him and us time and money.

I am going to leave you with a problem to solve for yourselves. In fact lets make it a little contest. I will extend your *Audio Basics* subscription for two issues, free, to each current subscriber that sends me a written solution of the problem, including logical method of solution, before November 20, 1982.

The problem is as follows: You have come home and turned on the system (on FM tuner) and find you have no output from the left speaker. You try the phono system and still no left channel output. You ignore the preliminary advice given in *Audio Basics* Issue Number Nine. You proceed to troubleshoot exactly as shown in our example #3 starting at the bottom of page 27. You find, after the first step, that when you turn on the system, there is a momentary loud hum from the right speaker, which then stops, and also there is no output from the left speaker. Now there is no sound from either channel on any source. You now cannot troubleshoot further (you think) with your own system, as both channels are dead. So, you borrow a friend's identical amplifier (its one of those DC to light jobs) and substitute it for yours in your system. When you turn the system on, there still is no sound at all. You take your friends amp back to his system and try again there. There is a momentary loud hum from both of his speakers, and then his system too is completely dead. You still don't give up. You have another friend with a similar system. You take your speaker systems over to his house. You connect them to his identical amplifier (keeping the left speaker on the left channel and the right speaker on the right channel). You turn on his system. There is no sound from the right speaker, but the left speaker still plays. Should you stop at this point? What has happened? What was the original problem? What is the problem now? What will happen if you swap the speaker wires again (left channel to right channel) on the system your speakers are currently connected to? List all of the defective components that now exist in the three systems (if any).

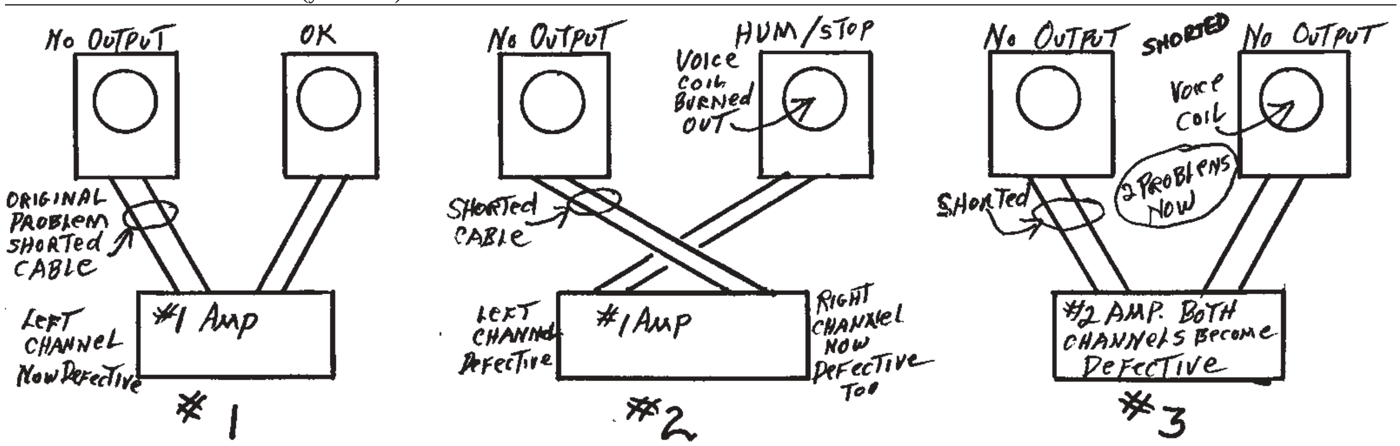
I will publish the solution next month, along with a "flow chart" to further help you with the logic of audio system troubleshooting. Our intent is to have you know more about the way your system works than the people who sold it to you do.

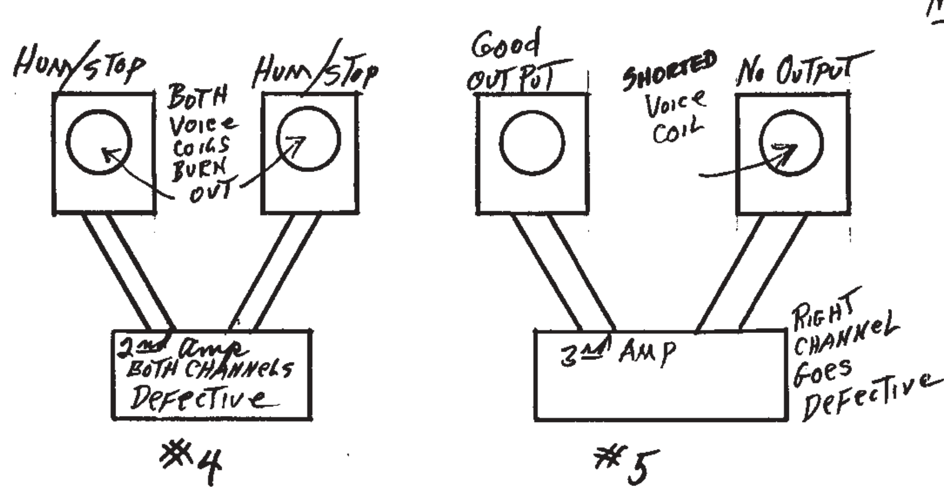
A friendly reminder and thank you. First, I appreciate your support shown by the large number of subscription renewals that have already come in from my note in Issue Nine. Again, you may renew your subscription for another year for $12.00 if you get your renewal check to us by December 1, 1982. After that date the price for 12 monthly issues will be $15.00. Please get your renewals in soon. Consider that perhaps *Audio Basics* might also be a unique and low priced Christmas Present for an audiophile friend. We will include a note to gift subscription recipients telling them who was thinking of them, if you give us adequate data. Let us know if you want the gift subscription to start with issue #1 (we consider the years past issues each to be non-dated and useful) or with the current issue. We will do it the way you want it, and, if the gift is to be for the back issues, we will send the packet to arrive before Christmas if you so instruct us to.

Frank Van Alstine

VOLUME ONE NUMBER ELEVEN
NOVEMBER, 1982

I will start this month with a further follow up on "Wonder Caps." Although we have heard not a word to date from International Audio Review,





Note: WITH ALL THE DAMAGE CAUSED BY ONE SHORTED CABLE, IT'S IRONIC THAT THIS SHORT PROTECTED THE LEFT SPEAKER BELONGING TO THE #1 SYSTEM.

George Reynolds

we are pleased to report the local factory representative of Reliable Capacitors did call on us regarding our request for information.

It seems that Reliable Capacitors is an established U.S. company that does make their own capacitors – not a jobber for an "off-shore" capacitor company as so many are. We were given a data book on their parts.

The particular capacitor that others are buying from Reliable and calling "Wonder Cap" is their PPMF series metalized polypropylene capacitor. We have samples on order and will report further.

One interesting characteristic of the "Wonder Cap" is that it is not a non-inductive capacitor. Thus some of its "sonic" characteristics may be the added inductance when this type of capacitor is substituted for a non-inductive capacitor. Keep in mind that added inductance is not desirable in a given circuit, and is destabilizing. Thus some of the golden ears may be hearing differences, and since "different must be better," may be assigning wonderful sonic qualities to added garbage. Again, this is to take nothing at all away from the real characteristics of Reliable Capacitors, each, we assume, will meet their specifications.

One other point, the PPMF series metalized polypropylene capacitors are physically large. They are high voltage (200 and 400 volt) and low capacitance (.001 to 10 μF). Thus they are suitable only for typical vacuum tube circuit applications, as are almost all plastic film capacitors. Inasmuch as vacuum tube circuits are basically "stone age" designs, I doubt if you will find many PPMF caps in our circuits.

Now, to return to Basic Audio System Troubleshooting, and for the results of last month's troubleshooting problem. I am a bit disappointed in my readers, to date (11-17-82) I have received only three entries, and of those, only two correct answers. The good news is that the two correct answers were done so thoroughly that I am going to reprint them here, as they are both done better than I could relate the solution. Dan Kolton (Michigan) sent in the detailed written response which follows, while George Reynolds (New York State) answered with a five step sketch diagraming the problem correctly. Refer to the sketch by Mr. Reynolds in reading Mr. Kolton's answer.

> "Dear Frank, Your little trouble shooting puzzler sure gave me fits for a while, but I think I've got it solved. I had to draw five diagrams to represent the five steps, and label components, color code wires, etc. , before I figured it out. The only two obvious things that struck me at first glance were that, a) the left speaker in system 1 was OK from start to finish, and, b) the right speaker in system 1 was OK before it was hooked up to the system 1 power amp left channel through its original speaker cable. I was thrown off by the "loud hum" for a while, because it suggested to me an ungrounded interconnect cable. However, I couldn't find a way to make that fit all of the steps." (Ed note: Many power amps make a "loud hum" as they blow up and go DC offset.)
>
> In any case, here are the answers to the questions you asked:
>
> - Should you stop at this point?
>
>   Only if you don't want to fry more speakers and power amps.
>
> - What has happened?
>
>   The left channel speaker cable in the original system has somehow been shorted. Maybe someone set a chair on your zillion strand litz wire. When you turned the system on, the short took out something in the power amplifier left channel, so that it was putting out full bore D.C., but the shorted cable protected the speaker from seeing it. Now, the full bore D.C. was fed through the perfectly good right channel speaker cable, and fried your right speaker. At the same time, the shorted left channel cable connected to the previously good right channel of the power amp makes it go the way of the left channel. You didn't hear it happen because of the short that made it happen.
>
>   Since you now have a shorted left speaker wire, and a fried (shorted) right speaker, you are going to make your friend's amp just as sick as yours, as soon as you hook it up and turn it on in your system. Then you are going to fry his speakers by putting his amp (now with two DC offset channels) back into his system. This is how I knew that the original speakers were not fused – a short was necessary to blow that channel of each power amplifier. You say you took the fuses out because the reviewers in *Absopure Loud* magazine said your speakers would sound better connected directly? How does it sound now?
>
>   Finally, you put your shorted right speaker in your second friend's (former friend?) system and fried the right channel of his power amp. You might as well swap speaker wires at the power amp on his system, so you can blow the other channel also (by connecting it to your shorted speaker) unless you think you've done enough damage already.
>
> - What was the original problem?
>
>   Shorted left channel speaker cable.
>
> - What is the problem now?
>
>   Shorted right speaker from your system has blown right channel of second friend's power amplifier.
>
> - What will happen if you swap speaker wires again?
>
>   You will blow your second friend's left channel power amp by connecting it to your shorted right speaker, and his blown right channel will fry your still good left speaker.
>
> * List all the defective components:
>
>   Your system: Left speaker cable shorted. Both channels of your power amp blown. Right speaker blown (shorted).
>
>   First friend's system: Both channels of his power amp blown (full DC offset). Both of his speakers fried.
>
>   Second friend's system: Right channel of his power amp blown.
>
> If you stop now, you won't blow the other channel, or blow the second friend's speakers by connecting them. I am not sure exactly what might cause the power amp to go to full DC output when forced to see a short, but I had a Crown DC-300 that fried some Bose 901's long ago when this happened because one side of a fuse holder had an intermittent connection (when open, it went to full DC). Obviously, Crown expected both fuses to blow if the amp was shorted. That would have been OK"

What can I say, obviously if you need a friend to help you with your hi-fi system, it would be nice to have Mr. Kolton or Mr. Reynolds to call on. A few other notes regarding the problem.

1. Obviously the problem could not have originated upstream of the shorted speaker wire as then the previous component would have "forced" distortion or noise from the original left speaker as it fried. This eliminates tuners, preamps, turntables, etc. from consideration.

2. The left channel speaker remained good as it was "protected" by the shorted speaker cable ahead of it. All the current went thru the short, which was in parallel with the speaker. Thus the original speaker never went "humm" – the sound of the power amp blowing, because all output of the amp was shunted thru the shorted cable. Note that defective amplifier channels did make subsequent speakers put out a momentary "loud humm" before they fried.

3 Many DC coupled amplifiers, when they fail, exhibit dead shorts from their power supplies, thru the shorted output transistors, into the speaker load. The resultant direct current, at several amps, along with lots of AC ripple as the supply is sucked down, does fry woofers. Again, many fried woofers do fail in a "shorted voice coil" mode, so that the woofer then presents a dead short to the next amp connected to it. The cycle can be endless.

4. Obviously all of the damage could have been prevented (except for the first blown amplifier channel) if the person had followed our basic rules in Issue #9 and had inspected all of his connections and cables before going further. It is much better to find the chair leg on the Litz wire first, rather than last.

5. A further hint was available to you if you had remembered what you have previously read in *Audio Basics*. We described part of this exact problem in *Audio Basics* Number Eight (constraint #2). Read it again, you saved me $800.00 (400 subscribers x $2.00).

Again, why this problem? Because it can and has happened and there is no need for it if you use a little logic with your system

How to Troubleshoot Your System, continued. If you notice a problem with your system first quickly notice what kind of a problem it is. Is it hiss, hum, distortion, or no sound at all? Next, note if it is a single channel problem (and which channel) or a problem common to both channels. Next, turn the system off! Think about the problem with the system shut down, not while it is busy frying your speakers.

Which source were you using when the problem occurred? Remember that too. Make a few quick notes as things can get confusing. Its a bit like playing "Clue" – It was Miss Peach in the Conservatory with the Rope.

No matter what the problem seemed to be, first go over every interconnect cable and speaker wire inch by inch. Make sure every cable is plugged in, the connections are solid, and that there are no stray wires shorting together or to a chassis in the wrong place. Do this with the system turned off! Chair legs and vacuum cleaners do short out speaker wires. Stiff interconnect cables do pop out. Fuses do let go of old age. Cartridge pins do corrode and go open. Stylus assemblies do get "dusted off" by the cleaning lady. Every service agency has heard, many times, the phone call from someone starting with, "We had a party last night and now my ——," or "I lent my stereo to a friend over the weekend and now ——." We get that phone call at least once a week!

Anyway, if you are suspicious of an interconnect cable or speaker wire, replace it. Replace any speaker wire that has a splice in it with one long enough to reach. Any Radio Shack store has a good supply of speaker cable or interconnect cables. Replace any multi-strand weave type speaker wire in any event (the "Cobra Cable" type) under the general principle that Murphy says that if they have not yet shorted out they are going to soon, and in the most damaging way possible (re: trouble shooting problem).

Check all user replaceable fuses. Note that there are two kinds of fuses. In general "slo-blo" fuses (ones with a "spring" inside the glass envelope) are used in applications in which there is a significant "surge" of current at turn on, such as main power amp fuses ahead of the power supply (the AC line fuse). These fuses are designed to withstand a substantial over-current for a limited time to allow, for example, a power supply to charge up. They fail when their current rating is exceeded on a constant basis for a few seconds. This kind of fuse is no protection at all for uses such as speaker line fuses. The slo-blo fuse will stand the over-current long enough for your voice coil or amplifier output transistors to fail first, thus protecting the slo-blo fuse. The good news is that you won't have to buy a new fuse, the bad news is that you will have to buy new speakers.

A "quick-blo" fuse is the type to be used to protect speakers and/or amplifier output stages. This type of fuse has a simple straight or slightly curly thin wire inside it. It will blow very quickly at the slightest overload. Obviously, this type of fuse won't work as the main AC line fuse in a power amplifier, the turn-on current surge as the power supply capacitors charge will blow it every time. Many amps have come in "for repair" because they "keep blowing fuses" when the only problem was the wrong kind of fuse was used. In checking fuses, do not overlook protection fuses built into the back panel of your speakers. We have had several Polk 10 speakers returned for "blown tweeters" when the only problem was an open 1 ampere tweeter fuse.

Regarding fuses, the following is a small chart showing the maximum continuous average power these common quick-blo fuses will handle, assuming an eight ohm load. If the load is four ohms, the power handling capacity of the speaker fuse is cut in half.

One Ampere	8 watts
Two Ampere	32 watts
Three Ampere	72 watts
Four Ampere	128 watts
Five Ampere	200 watts

NOTE: Power = current squared times resistance = I^2*R

Power = voltage squared divided by resistance = V^2/R

Note that it is absolutely useless to "overfuse" a speaker or amplifier. The largest speaker fuse that should be used with a 75 watt class amp is 3 ampere, for example. You may assume that a two way system should not be fused heavier than 2 ampere, and a large three way system not heavier than 4 ampere, in the absence of data from the speaker maker.

If, when checking fuses in your preliminary inspection of your system, you find that a speaker fuse or an amplifier output fuse has blown, replace it with a one ampere quick-blo fuse for further testing purposes. The one amp fuse will hold for music played at low levels, but will blow quickly if there is an amplifier or speaker fault, protecting the equipment better than a heavier fuse will do. If, in the troubleshooting problem contest of last month, the user had done nothing but install a one ampere quick-blo fuse in each speaker line before proceeding, he would have saved all the rest of the equipment from damage.

Anyway, you should now have checked your interconnects and fuses, and know if any connections were loose, any fuses were blown, and you should have a few notes to yourself telling which channel had the problem, what kind of noise was being produced, and what source was used. If you found any blown speaker or output fuses, they have been replaced with one ampere quick-blos, while you have marked and saved the blown fuses (see Issue #9). Power supply fuses should be replaced with the same value slo-blo types.

Now turn on the system again, keeping the volume control down. If any fuse quickly blows again you can assume the following:

A If a slo-blo power supply fuse fails, then there is a defect in that piece of equipment.

B If a quick-blo speaker fuse or output fuse fails, then there is either a defect in the power amp output section, a shorted speaker wire (if the fuse is ahead of the cable – it should be), or a shorted speaker. Assuming you have kept your volume control all the way down, it is possible, but unlikely, that the problem may be in the audio output section of your preamp. It cannot be ahead of the volume control (any source) as the signal from the source is not being passed to the amp and speakers with the volume off.

Turn the system off, replace the blown one amp quick-blo fuse with another one, and connect your left channel speaker wires to the right amplifier output, and the right channel speaker wires to the left channel amplifier output, as in sketch #2.

Turn the system on again, keeping the volume down. If the fuse is at the back panel of the amp, and the same fuse quickly blows again, the problem is in the amplifier output circuits or the preamp output circuits. This can be "pinned down" by turning system off, replacing fuse once more, and swapping the interconnect cables from preamp to amplifier, at the preamp end only, while changing nothing else. Again turn on system. If the same fuse blows again, the problem is isolated to the power amp output section. If the other channel amp fuse now blows, the problem is the preamp output section, which is dumping a "garbage" signal into whatever side of the power amp it is connected to.

If the fuse is at the speaker, and the same fuse blows again, the problem must be a shorted speaker, as the speaker is now connected to the other (working) amplifier channel. If the fuse on the other speaker now blows, then both speakers are OK (assuming you didn't overfuse them), and the problem must be the amp or preamp which can be isolated as described in the paragraph above.

If the fuse is at the amplifier, and the other channel fuse blows when the speaker wires are reversed as in sketch #2, then the problem is a shorted speaker or speaker cable. You can then reverse the speaker wires at the speaker ends to determine if the problem is the speaker or the speaker wire.

To continue, let us assume that you are lucky and no fuses blew when you turned the system back on. Make sure the system is switched to the same source with which you first noticed the problem. Slowly advance the volume control to a low playing level. If the problem is now gone (especially if it had been no signal or a loud hum) then it most likely had been an open or badly grounded connection which you have now cured. Continue to use the system and hope the problem does not reoccur. Save your notes. Date them. This can save you money if you have an intermittent problem and the unit in question is under warranty now, but the warranty is over before you pin down the problem. If you have evidence that the problem existed during the warranty period, the manufacturer may extend the warranty coverage.

If the problem is still there (no blown fuses, but hum, hiss, noise, distortion, or silence from one or both channels), turn the volume all the way down.

A. Does the problem (especially hum, hiss, or distortion) "follow" the volume control setting? Louder when volume is advanced, quieter when volume is turned down?

1. If yes, then the problem is ahead of the volume control, eliminating the speakers, amplifier, and output section of the preamp from consideration. (Two exceptions – if the problem is a "scraping – buzz" in time with the "beat" of the music, it is probably a speaker voice coil defective and rubbing against its magnet – swap speakers channel to channel and if the problem "follows" the speaker, this confirms it. If the problem is distortion that gets worse as the volume is increased then swap the interconnects from the preamp to power amp at the preamp end. If the distortion remains in the same speaker then it is an amplifier problem. If it changes to the other speaker, then it is not the power amp or speakers.)

2. Try another source (if the problem was with the FM tuner, try another station – if the problem is only with one station then the problem is at the FM station – they broadcast a bung signal with regularity, or your antenna is picking up multi-path reflections from that station – readjust or replace FM antenna). If the problem goes away on all other sources (tuner, tape, phono, etc.) and is there only on one source, then the problem is isolated to that signal source. If the problem is on all sources, and you have eliminated the amp and speakers as in the previous paragraph, then the problem is in the output section of your preamp.

3. If the problem is there only with the phono as a source, first try another record – some records are badly distorted or noisy. If the problem exists with all records, swap the phono leads to the preamp at the preamp. If the problem changes channels (switches from left to the right speaker), then the problem is in the phono player itself. If the problem does not change channels, it is in the phono RIAA preamp section of your preamplifier.

4. If the problem has been isolated to the phono player itself, swap the headshell wires from the left to the right channel of the phono cartridge. If the problem then changes channels it is a defective cartridge. If the cartridge has a user replaceable stylus assembly, try removing and reinstalling it. Sometimes internal contacts go open and will be "wiped" clean in this operation. (Don't overlook something as simple as a dirty stylus). If this doesn't fix it, try a replacement stylus. Caution – if your cartridge is old, always keep in mind that you may be able to buy a much better complete new cartridge for less than the cost of a replacement stylus for your old cartridge. Major improvements have been made in phono cartridges over the years, and many inexpensive cartridges made now outperform even the top of the line models of a few years ago.

   Phono cartridge distortion can also occur if the cartridge is tracking too heavy or light. Buy a good stylus gauge such as made by Shure or AR. Set your cartridge to track at the heavy end of the manufacturer's recommended range. (If the "spec." is "tracks at from 0.7 to 1.5 grams," set it at 1.4 grams for starters, the 0.7 gram is wishful thinking). You can also make a very useful anti-skate setting tool with a quick trip to a local glass shop. Have them cut you a record sized circle of plexi-glass, complete with accurate center hole. This blank disc will show up large anti-skate errors when played instead of a record. If the arm pulls in, the anti-skate is set low, if it pulls out, there is too much anti-skate. Adjust anti-skate so the arm neither moves in or out when set on the turning blank disc.

Out of space again, Troubleshooting will be continued next month. Remember, get your subscription renewal in before the end of November if you care to renew at the $12.00/year rate. After 12-01-82 the price goes to $15.00/year. Have a good Thanksgiving.

Frank Van Alstine

VOLUME ONE NUMBER TWELVE DECEMBER, 1982

Continuing follow-up on "magic capacitors" – still no data of any kind or response from *International Audio Review* regarding "Wonder Caps." I "wonder" how they expect to sell them? No samples from Reliable Capacitors either. I "wonder" if it has been worth the effort – it makes one "wonder" if they really are all that wonderful?

Now for a follow up on the Troubleshooting Contest. Three more correct entries came in before the deadline from: Emerson Hawley of New York, Perry Smith of Wisconsin, and Z. Z. Hugus of North Carolina. Another correct response came in from Gary Phillips of Michigan, but too late to qualify for the subscription extension. Now for the interesting part – another troubleshooting contest.

Mr. Hugus submitted an absolutely correct solution that was completely different from the solution we had in mind and described last month. Mr. Hugus' solution took into consideration every constraint given in the problem, and explained every annoying sonic manifestation I described, but in his solution no equipment was damaged at all. One hint – the "audiophiles" described in Mr. Hugus' solution were Absolute Klutzes. The contest – give me Mr. Hugus' solution before January 10, 1982, in writing for a free extension of two issues to your subscription of *Audio Basics*. Mr. Hugus need not reply – he wins automatically – and no fair calling Mr. Hugus. You wouldn't do that, would you?

If you find a red "your subscription is ending" sheet in with your copy of *Audio Basics*, then this is, according to our records, your last issue. We will include that simple notice from now on, rather than "begging" in the body of the copy. This start-up year was a bit special, as we had most subscriptions ending at the same time. We certainly appreciate the support you have given us – our renewals exceed 200 already.

I ask one favor from those of you who have chosen to not continue with *Audio Basics* – please write to me and let me know what we are not doing right. What did you expect that we did not deliver? What should we do (within the context of subscription cost vs. our choice of not operating with a government subsidy of our postage costs) to make *Audio Basics* interesting and valuable enough for you? Have you actually tried any of our suggestions? Some of those that have, have reported to us that any one of them was of more value in their system than the subscription cost. I do take pride in reminding you that I have kept one promise – I promised 12 monthly issues – you got them. Has any other "underground magazine" kept that promise?

Last month we left off with phono cartridge distortion. This is a good place to go thru one common lament we hear regarding phono cartridge tracking, specifically – "my cartridge will not track all the cannon shots on the Telarc digital recording of the Tchaikovsky "1812 Overture."

The answer to this lament is not what you expect or want to hear - the answer is, "it doesn't matter." This is an unusual case of a dichotomy between what you want your system to do, and what it can – and in the final analysis, should do. The reality of the situation is that it is not possible for any home system to play the cannon shots, and if the system could play them, you would not want to play the record on your home system in any event. Let me explain what is happening.

If we play the Telarc 1812 recording on our system with the phono arm cartridges set up very carefully so the cartridge does not mis-track, and if we set the playback level so that the normal quiet parts of the orchestral passages are at a minimum soft, but audible level for my quiet and very dead listening room, then the measured peak levels (using a fast Tektronix scope to monitor the preamp output) on the cannon shots exceed 8 volts! That is correct – eight volts peak signal into the power amplifier inputs. Do you know what this means?

Consider that with a typical high powered amplifier with "normal" voltage gain, a two volt peak input signal drives a 200 watt amplifier to full power. Consider also that power is a voltage squared function, not a linear function. Thus, to avoid clipping your power amplifier with an input signal in excess of eight volts, you must have a power amp capable of more than 3200 watts per channel! That's right, run down to your local audio salon and bring home a 7000 watt power amplifier (and while you are at it, call your electric company, as, all things being equal, you are going to need a 50 ampere at 120 volt AC line for the amplifier). And finally you really will need Monster Cable as your speaker wire will need to handle in excess of 20 amperes of continuous current per channel. Have your fire extinguisher handy!

Now, when you play the Telarc 1812 cannons, your power amplifier will handle them full power, without clipping. Congratulations, you just disintegrated your speaker systems! At this point you had better run out and purchase speaker systems that will handle 3200+ watts each too. A few handy hints – this may be another good use for Monster Cable – to wind the voice coils with! Consider that a dandy electric space heater draws about 1200 watts. Thus the heat your speakers must dissipate will be about the same as running 6 electric space heaters at the same time – helpful hint – don't play the record on a hot day, save it for when its cold out and take a real load off of your furnace. To save time, we will assume that you now have a set of speakers that will handle 3200+ watts with the same acoustic efficiency as the ones you previously destroyed.

Now try playing the record again in a system that does have adequate amplifier power and does have speakers that will handle the nearly 7000 watts of power. Now what happens? Congratulations – you're dead! You have had the acoustical output of a live military cannon go off in your very own living room. When they clear away the rubble and find your shattered body, perhaps the remnants of the record jacket will help the rescue squad understand that it wasn't a gas explosion.

You are still not sure about this? Lets look at it another way. We can gain acoustic output by installing more amplifiers and speakers. Thus to have the equivalent acoustic output of an unclipped 4 volt peak signal, all we need are four of our Transcendence 400 amplifiers driving 4 pairs (8) B&W 801F speaker systems with a 2 volt peak input signal to each channel. Continuing, since our goal is to have unclipped acoustical output equivalent to that 8 volt peak input signal, all we need in our living room are 16 Transcendence 400 amplifiers (inputs paralleled) driving 32 B&W 801F speakers with the preamp turned down so that the peak output of the preamp is 2 volts. Of course now you won't be able to hear the soft passages, but that really doesn't matter, as when the cannons go off, you will, at best, be permanently deaf anyway, so you won't be able to hear further soft passages at all. Or, one could judge that 16 Transcendence 400 amps + 32 B&W 801F speakers would been just fine for handling a Plasmatics concert in a large auditorium. Think about compacting that total acoustic output down into your living room – isn't nice to think about, is it?

Actually, the way the Telarc 1812 record plays on your home system, using real world equipment, is only an indication of how your system clips and overloads – it cannot be played cleanly. A very stable system will clip with less gross distortion and with less chance of bottoming woofers, going offset and frying speakers. Is the record a "good test" of your system? Is driving your car into a brick wall at 40 mph a "good test" of the car's performance? Enough said.

However, I think this does bring up another question that should be answered now, further digressing from system troubleshooting, namely HOW MUCH AMPLIFIER POWER DO YOU NEED, NOW, AND WHEN TRUE DIGITAL SOURCE MATERIAL IS AVAILABLE? From our standpoint, as amplifier designers, we need to consider this question too – what should be our design goal? Lots more power, more "dynamic headroom" (note that dynamic headroom is an inverse correlation with the quality of the amplifier power supply – an amplifier with a perfect power supply would have zero "dynamic headroom" – more about that later) or whatever?

What should our design goal be for future products? How much power do you (or will you) need? Let us consider the following constraints:

Constraint #1. There is an upper limit to the level of acoustical energy that you can physically tolerate before your ears are damaged. It would be pointless (and probably irresponsible) to supply audio amplifiers capable of output in a home environment such as required for the Telarc 1812 record as described above. There is some debate among experts as to the acoustical levels that cause hearing damage, but I suspect that we need never supply an amplifier with more power than that required to drive a typical inefficient speaker to more than 120 dB peak levels.

Constraint #2. All other things kept equal, the cost to manufacture a higher powered amplifier skyrockets up at an exponential rate. If we wish to provide twice the voltage swing (four times the power – 800 watts/channel and a 6 dB increase in acoustic output, for example) then our amplifier must use 32 power mos-fet devices, each with a 300 volt rating (no such part is available, thus we must series-parallel four devices to replace each single device to maintain reliability and end up with 4 x 32 or 128 output mos-fets!). This is going to take a very big heat sink! Next we are going to need to scale up the power supply. Where a 60,000 µF supply at + and - 80 volts is just fine for the 200+ watt/channel Transcendence 400, for our "high powered" amplifier we must have a 120,000 µF power supply at + and - 160 volts. This presents another problem – nobody makes 160 volt rated power supply capacitors with any real amount of capacitance. About the best we see in our Mallory catalogue is 7000 µF at 200 volts in a case 3" in diameter and 6" high. We will need 18 of these animals, and that will require about 6" x 27" of "floor space" – sorry folks, we are not going to give you a "rack mount" unit, the rack will be too small to fit our chassis. We have now screwed up one of our main design requirements – a very compact size power supply located very close to the output devices, connected to reduce lead inductance as much as possible. With power supply caps this big and bulky, we cannot put the supply close to anything. After replacing our 1000 watt rated Transcendence 400 power transformer with a 4000 watt rated one I suspect our amplifier is going to get a bit heavy – try about 200 pounds! After we scale our drive circuit up too, using devices with twice the operating voltage but with equal beta and linearity (there are not any) driving twice as far with twice the gate current, we are going to have an expensive circuit. This is before we double the slew rate of the whole mess to maintain the same power-bandwidth of the 200 watt amplifier. I doubt if we could do it for under $10,000 per amplifier. Are you willing to pay for it? Please note that we are talking about real power into a loudspeaker load under varying transient conditions and assuming a combined resistive-inductive-capacitive load. We are not discussing an amplifier that makes a short term peak IHF rated power into a non-inductive 8 ohm test load only, when tested with a simple low-distortion sine wave, single frequency at a time. “Paper” watts are easy to make cheap, the problem being that “paper” watts won’t drive a loudspeaker, only a specification sheet.

Observation #1. Actual live acoustical musical performances “sound better” than even the best audio equipment.

Observation #2. Even the best high fidelity systems do not sound too wonderful at loud levels. Where one can enjoy the sound of a good jazz group in a small club even when the sound (unamplified) is too loud to talk over, a home hi-fi system at the same levels is pretty groady.

Observation #3. No audio equipment deserves a “god’s gift to audiophiles 5 star wonderful” rating in any golden ear underground publication. My 12 year old son, playing a few seconds of pretty good music on his trombone in my listening room, completely discredits any attempt by an esoteric reviewer at claiming any given piece of hi-fi equipment really approaches true linearity. To be blunt, the best any of us are doing now is making equipment that doesn’t sound “bad.” Reality is still a ways away.

Observation #4. If the coming digital source material does actually deliver its claimed 90 dB dynamic range, and if we assume we really don’t want to be exposed to peak levels over 120 dB, then scaling that dynamic range back down tells us the lowest levels on the digital players will be at 30 dB, if the system is set to go no louder than 120 dB. This presents another limitation – our room isn’t that quiet! If you are getting your listening room’s ambient noise level down to 30 dB than you are doing extraordinarily well! Most of us cannot use 90 dB of dynamic range. Actually, in a typical home system, the producers of digital material (laser read disks or VCR tapes) will be doing us a favor if they actually use about 15 dB of compression! If they don’t, and we don’t want the loud passages to be at ear shattering levels, then the soft parts will be lost in the ambient room noise! Again, 90 dB of dynamic range may be too much dynamic range for home use.

Observation #5. We have had the opportunity to use the Transcendence 400 (200+ real watts/channel) at Soundstream in Salt Lake City, driving Infinity 4.5 speakers full range (the most inefficient speakers I know of and a one ohm load at some frequencies) in Soundstream’s mastering studio (a large dead room, bigger than almost all home environments) with their master digital tapes as a source. Note that the Soundstream masters (the system used by Telarc) have a 50,000 Hz sampling rate and is a true 16 bit system with greater dynamic range than the coming VCR digital tapes and digital audio disc. The Transcendence 400 was adequate to drive the pair of Infinity 4.5’s to overpowering acoustic levels without overload. Yes, their system did distort (break up) on the 1812 cannons – they had spliced about 20 cannon shots in a row just for fun. However, we suspect it was the speakers bottoming that was the main problem – not the amp in gross clipping. Of course, if the speakers had not bottomed the amp certainly would have been heard clipping!

Now lets try to tie together these constraints and observations into a series of conclusions:

Conclusion #1. If we do build you a much more powerful amplifier, you are not going to be able to afford to purchase it and we will go broke designing and building a product we cannot sell.

Conclusion #2. If you do purchase our 5000 to 10,000 watt amplifier, it is going to destroy your speaker systems.

Conclusion #3. If you purchase speakers that can handle this kind of power, you are going to destroy your ears.

Conclusion #4. We now build an amplifier (Transcendence 400) that is adequately powerful enough for the most inefficient speakers in a large home environment on all but the most ridiculous digital source material. Further, if your speakers are more efficient and your room not huge, the power required can be scaled down appropriately.

Conclusion #5. Inasmuch as it will be necessary to compress digital records to scale the dynamic range down to that useful in a home situation so that the soft passages are not buried in the room noise and the loud passages do not destroy your ears, the necessary amplifier power can be scaled down accordingly.

Conclusion #6. (Actually an observation) The real advantage of the coming digital material is not its increased dynamic range potential, but its linearity – noise is gone, wear is gone, all the non-linearities associated with the multi-copy process now needed from master tape to record including the gross distortions in the tape recorder, the cutter amp, the mechanical master record cutter, all the plating processes in making the stamper, the pressing process, the vinyl noise of the finished plastic record, and the gross non-linearities of the crude mechanical phono cartridge-arm-turntable system, the difficulties of the RIAA equalized phono preamplifier, all eliminated in one swell foop. Your digital disc will be, (assuming reasonable process quality control) not a copy, but a clone of the master digital recording. This dear reader is the advantage of digital – finally a non-distorted source.

Conclusion #7. Inasmuch as “live” still sounds much more like live than even the best audio equipment, even when the audio equipment is run well within its power limitations, we can judge that the major problem is inadequate linearity in the audio system, not inadequate power.

THEREFORE: Our future design goals are aimed towards engineering and building audio amplifiers with substantially better internal linearity under real world operating conditions, not towards building “more powerful” audio amplifiers. We would suggest that the obvious fact that hi-fi systems do not yet approach the actual sound of live, flatly contradicts the editorial position taken by major advertising supported audio publications claiming that “all audio amplifiers are wonderful – near perfection – and essentially identical.”

We understand their problems. Given the very limited scope of their objective test procedures, all amplifiers do indeed measure nearly the same. All automobiles have nearly the same gas mileage too, if measured starting at the Eisenhower Tunnel in Colorado and driving downhill to Denver. What the major magazines overlook is that the total (and audible) non-linearities of an amplifier are the sum total of all the internal non-linearities when operating in real world use conditions, not just the non-linearities measured under narrow band IHF standards. It appears that the subjective listening policies of the major magazines is an attempt to sanction their limited testing procedures, and to pacify their advertisers (the real source of their income). Thus contradictions abound in their writing. Unfortunately, some who have the technical ability to do better (the B.A.S. types) seem only to be interested in measuring to IHF standards, but with finer resolution. Since they seem to wear the same set of “blinders” as the majors, their conclusions have yet to have any correlation with reality.

The “golden ears” are off in another “never-never land.” They have flatly rejected technology and are depending on subjective listening only to “cure all.” Although I would never deny that a trained audiophile can hear problems in any audio system, in the absence of any repeatable, objective, and verifiable set of standards, the audiophile cannot do better than the vague “I like it better, or I don’t like it as much” judgment. He cannot objectively describe it as “good” or “bad,” and cannot accurately assign reasons for things he does not like. To identify and eliminate objective non-linearities requires a systematic and scientific documentation of what non-linearities exist. And that dear reader, is our goal and methodology.

Frank Van Alstine